Sonus SBC 5XX0 5.0.0R1 IOT CUCM 10.5 CenturyLink SIP Trunk Application Notes

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Document Overview

This document provides a configuration guide for Sonus SBC 5XX0 Series (Session Border Controller) when connecting to Cisco Unified Communications Manager 10.5 (CUCM 10.5) and CenturyLink SIP Trunk.

This configuration guide supports features given in Cicso UCM configuration guide.

- For additional information on CUCM 10.5, please visit http://www.cisco.com/c/en/us/products/collateral/unified-communications/unified-communications-manager-callmanager/datasheet-c78-731839.html
- For additional information on Sonus SBC 5XX0 Series, please visit http://sonus.net.

Introduction

The interoperability compliance testing focuses on verifying various inbound and outbound calls flows between Sonus SBC 5XX0 Series and CUCM 10.5.

Audience

This technical document is intended for telecommunication engineers with the purpose of configuring the Sonus SBC 5XX0 series aspects of the CenturyLink SIP trunk group together with the CUCM 10.5 . There will be steps that require navigating a third-party and Sonus SBC Web browser user interface, Embedded Management Application (EMA). Understanding the basic concepts for IP/Routing, SIP/RTP and TLS are also necessary to complete the configuration and for troubleshooting, if necessary.

> This configuration guide is offered as a convenience to Sonus customers. The specifications and information regarding the product in this guide are subject to change without notice. All statements, information, and recommendations in this guide are believed to be accurate but are presented without warranty of any kind, express or implied, and are provided "AS IS". Users must take full responsibility for the application of the specifications and information in this guide.

Requirements

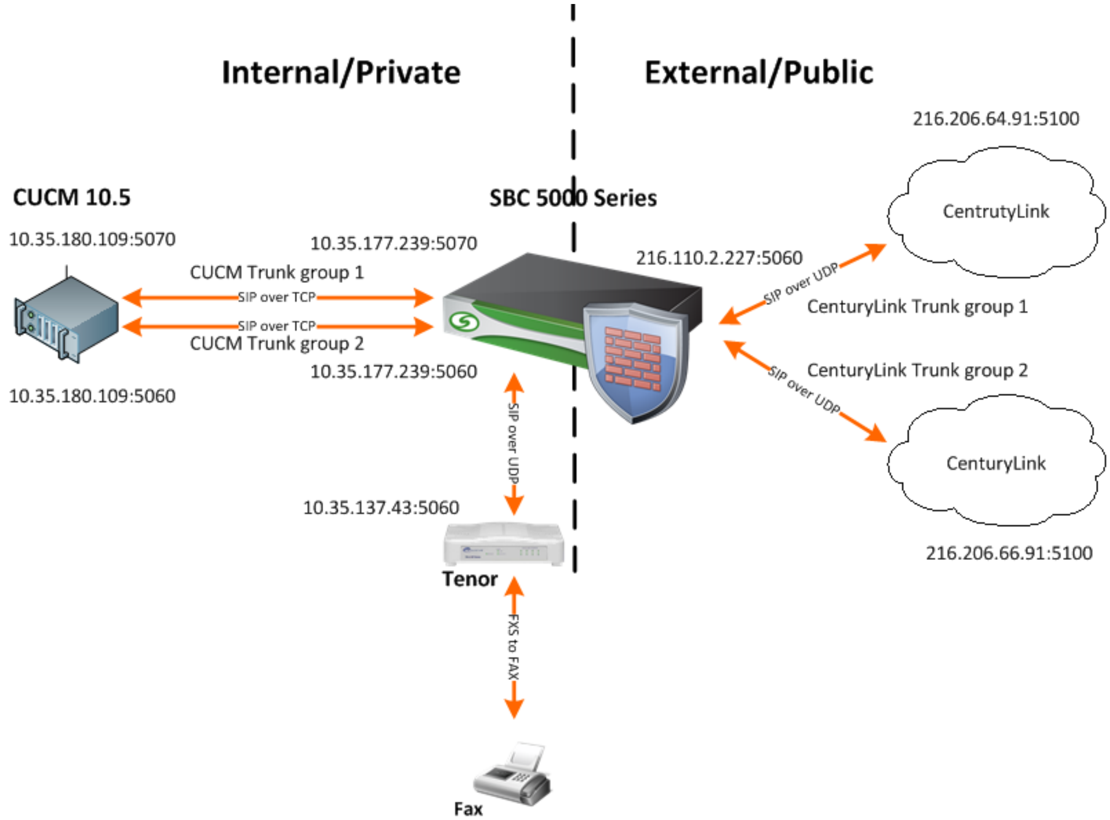
The following equipment and software were used for the sample configuration provided:

	Equipment	Version
Sonus Networks	SBC 5XXX	V05.00.00-R001
	BMC	V02.11.00
	BIOS	V02.06.00
	ConnexIP OS	V03.00.00-R001
	SonusDB	V05.00.00-R001
	EMA	V05.00.00-R001
Third-Party Equipment	Cisco Unified Communications Manager	10.5.1.11005-2
	Cisco IP Phone 7942	9.3.1.57

Reference Configuration

The following reference configuration shows connectivity between third-party and Sonus SBC 5XX0.

Figure 1: Reference Configuration



Support

For any questions regarding this document or the content herein, please contact your maintenance and support provider.

Third-party Product Features

The testing was executed with the CenturyLink test plan, and the following features were tested:

- Basic originated and terminated calls
- Basic inbound/outbound call
- Hold and resume
- Call Forwarding Unconditional
- FAX
- DTMF
- Dual Trunk
- Features codes

Verify License

No special licensing required

CUCM 10.5 Configuration

The following new configurations are included in this section:

1. Security Profile

2. SIP Profile
3. SIP Trunk
4. Route Group
5. Route List
6. Route Pattern

1.Security Profile

Select **System > Security > SIP Trunk Security Profile**

Figure 2: Security Profile First Trunk

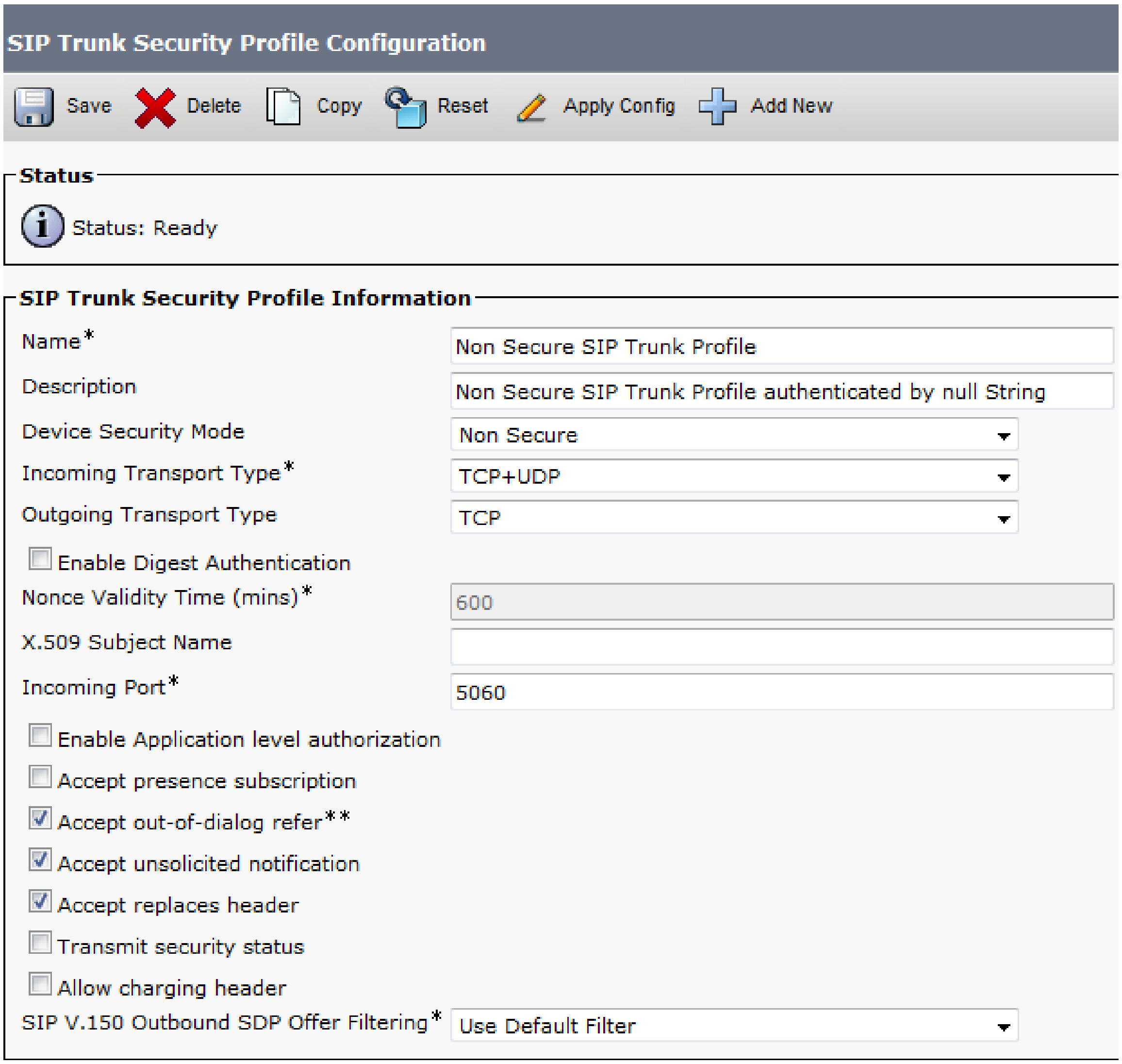
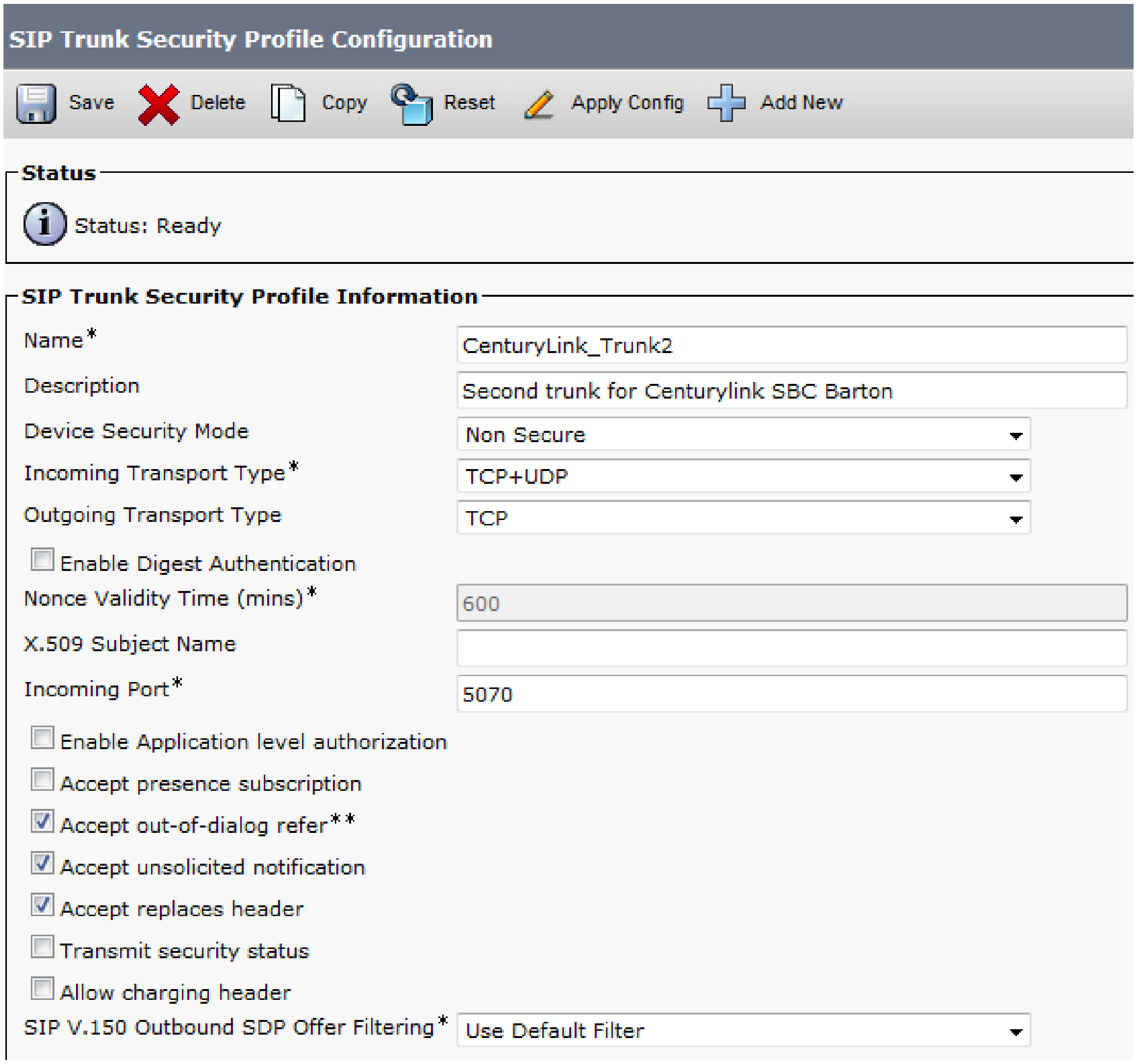


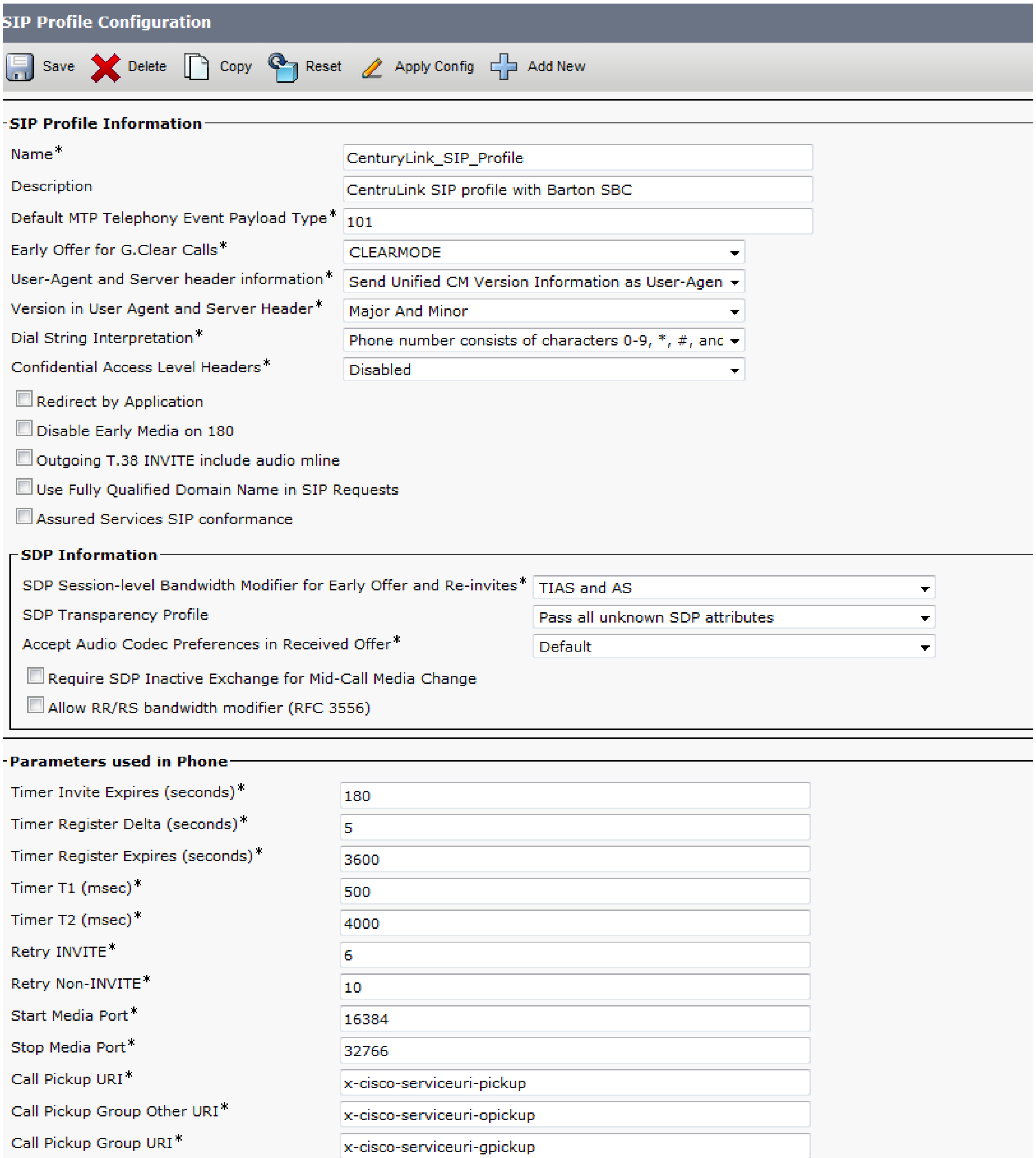
Figure 3: Security Profile Second Trunk



2.SIP Profile

Select **Device > Device Settings > SIP Profile**

Figure 4: SIP Profile



SIP Profile Configuration

Save Delete Copy Reset Apply Config Add New

SIP Profile Information

Name*	CenturyLink_SIP_Profile
Description	CentruLink SIP profile with Barton SBC
Default MTP Telephony Event Payload Type*	101
Early Offer for G.Clear Calls*	CLEARMODE
User-Agent and Server header information*	Send Unified CM Version Information as User-Agen
Version in User Agent and Server Header*	Major And Minor
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, anc
Confidential Access Level Headers*	Disabled

- [ ] Redirect by Application
- [ ] Disable Early Media on 180
- [ ] Outgoing T.38 INVITE include audio mline
- [ ] Use Fully Qualified Domain Name in SIP Requests
- [ ] Assured Services SIP conformance

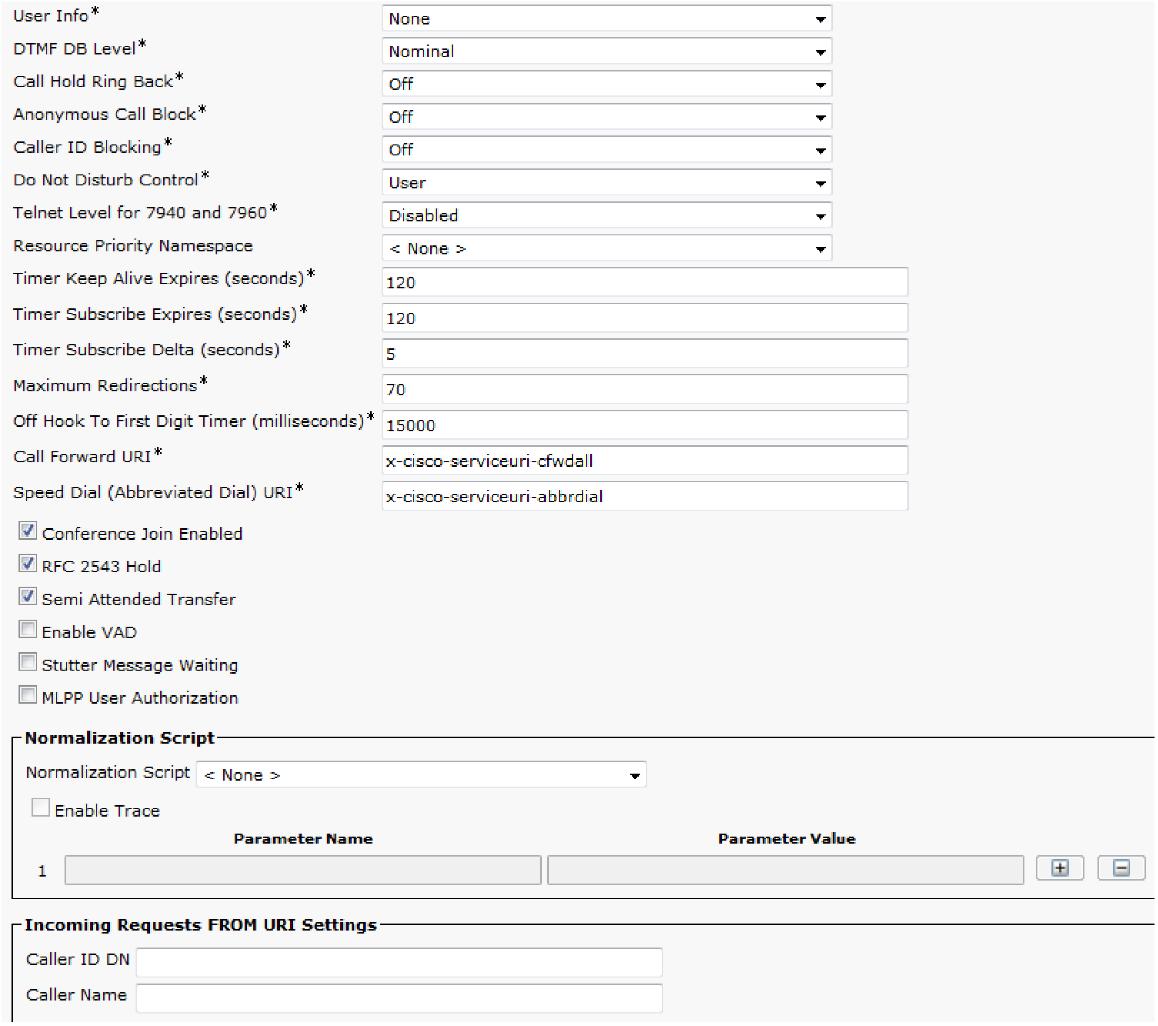
SDP Information

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
SDP Transparency Profile	Pass all unknown SDP attributes
Accept Audio Codec Preferences in Received Offer*	Default

- [ ] Require SDP Inactive Exchange for Mid-Call Media Change
- [ ] Allow RR/RS bandwidth modifier (RFC 3556)

Parameters used in Phone

Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup



3.SIP Trunk

Select **Device > Trunk > Add New**

Figure 5: First SIP Trunk

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Barton_SIP_Trunk
Description	CenturyLink IOT with Barton SBC
Device Pool*	711_DP
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0

- [ ] Media Termination Point Required
- [x] Retry Video Call as Audio
- [ ] Path Replacement Support
- [ ] Transmit UTF-8 for Calling Party Name
- [ ] Transmit UTF-8 Names in QSIG APDU
- [ ] Unattended Port
- [ ] SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default
Use Trusted Relay Point*	Default

- [x] PSTN Access
- [ ] Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile	< None >

MLPP and Confidential Access Level Information

MLPP Domain	< None >
Confidential Access Mode	< None >
Confidential Access Level	< None >

Outbound Calls

Called Party Transformation CSS	FullCSS

- [x] Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS	< None >

- [x] Use Device Pool Calling Party Transformation CSS

Calling Party Selection*	Originator
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling and Connected Party Info Format*	Deliver DN only in connected party

- [ ] Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS	< None >

- [x] Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN	
Caller Name	

- [ ] Maintain Original Caller ID DN and Caller Name in Identity Headers

SIP Information

Destination

- [ ] Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status
1*	10.35.177.239		5060	N/A

MTP Preferred Originating Codec*	711ulaw
BLF Presence Group*	Standard Presence group
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	CenturyLink_SIP_Profile View Details
DTMF Signaling Method*	RFC 2833

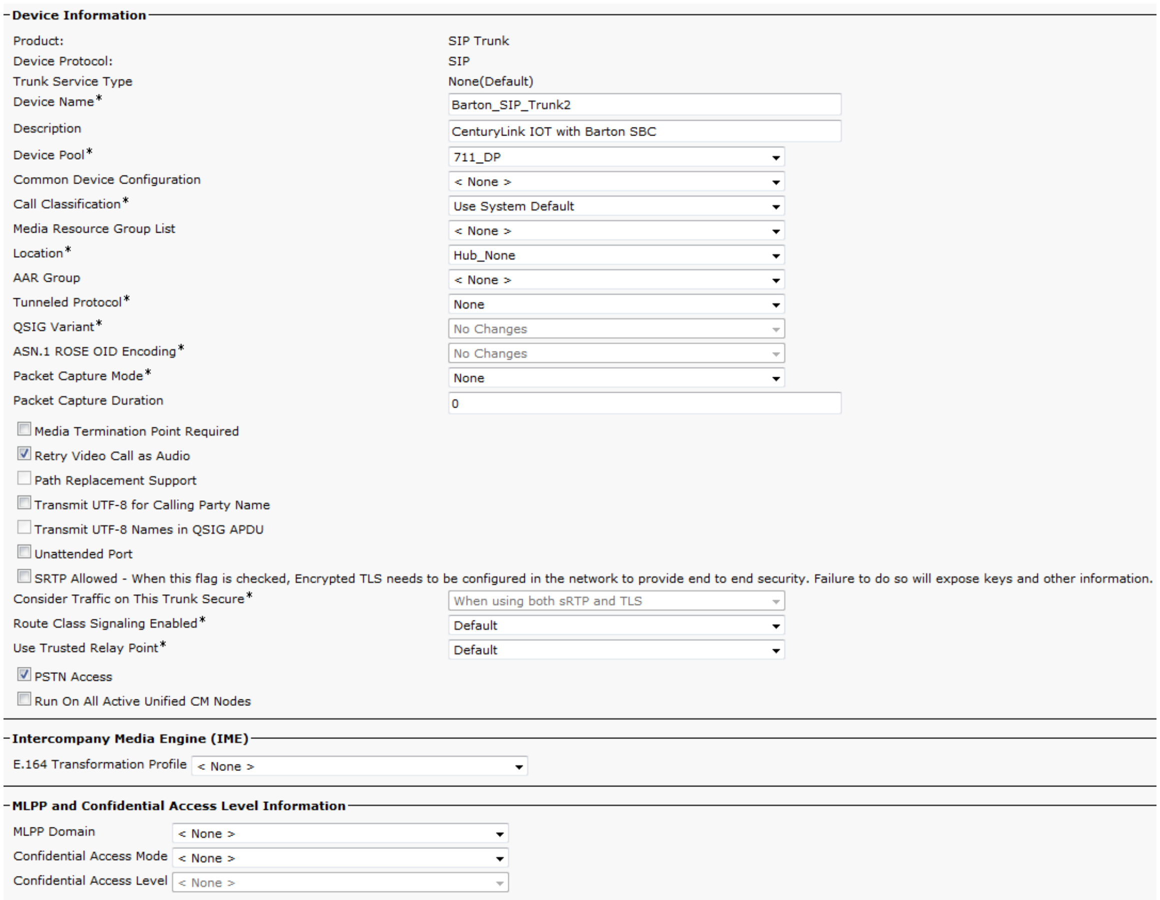
Normalization Script

Normalization Script	< None >

- [ ] Enable Trace

	Parameter Name	Parameter Value
1		

Figure 6: Second SIP Trunk



Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Barton_SIP_Trunk2
Description	CenturyLink IOT with Barton SBC
Device Pool*	711_DP
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0

- [ ] Media Termination Point Required
- [x] Retry Video Call as Audio
- [ ] Path Replacement Support
- [ ] Transmit UTF-8 for Calling Party Name
- [ ] Transmit UTF-8 Names in QSIG APDU
- [ ] Unattended Port
- [ ] SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default
Use Trusted Relay Point*	Default

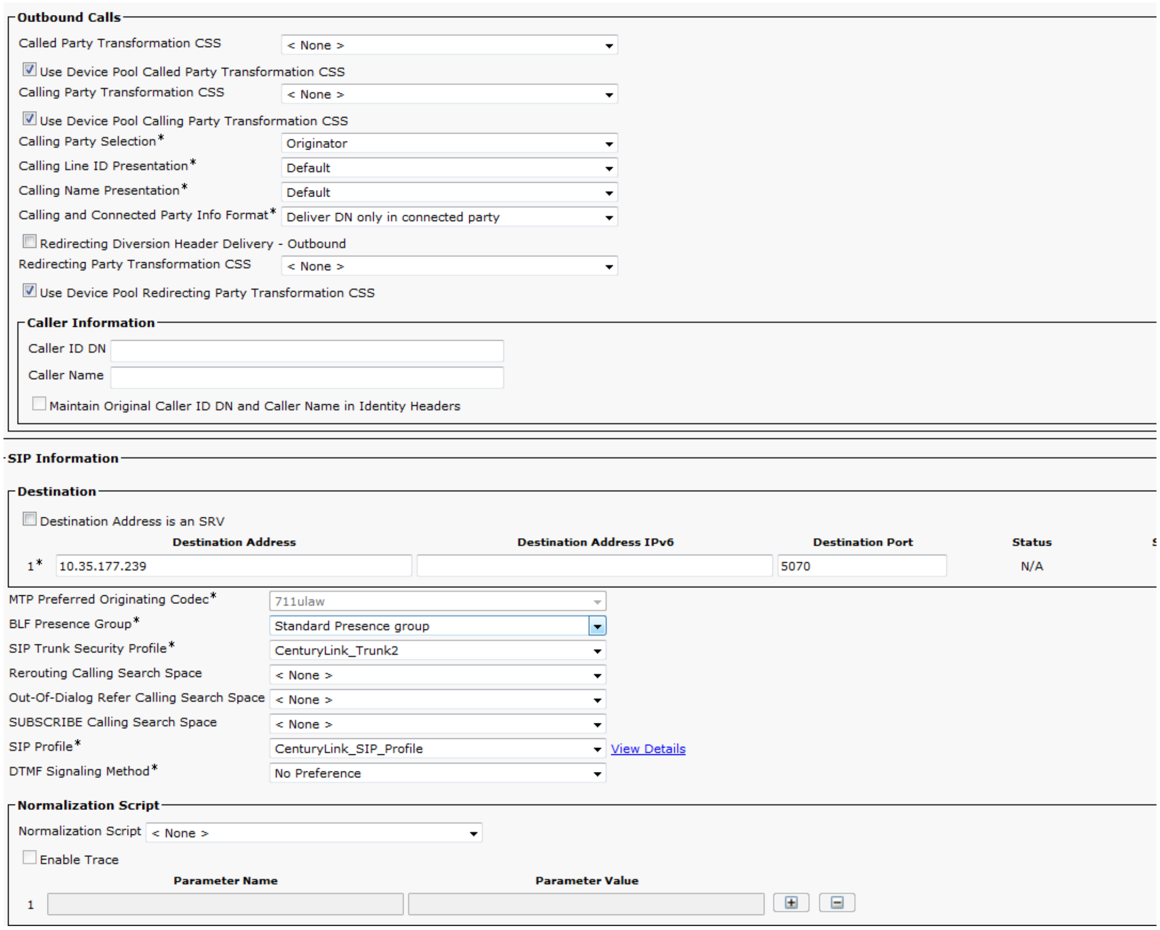
- [x] PSTN Access
- [ ] Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile	< None >

MLPP and Confidential Access Level Information

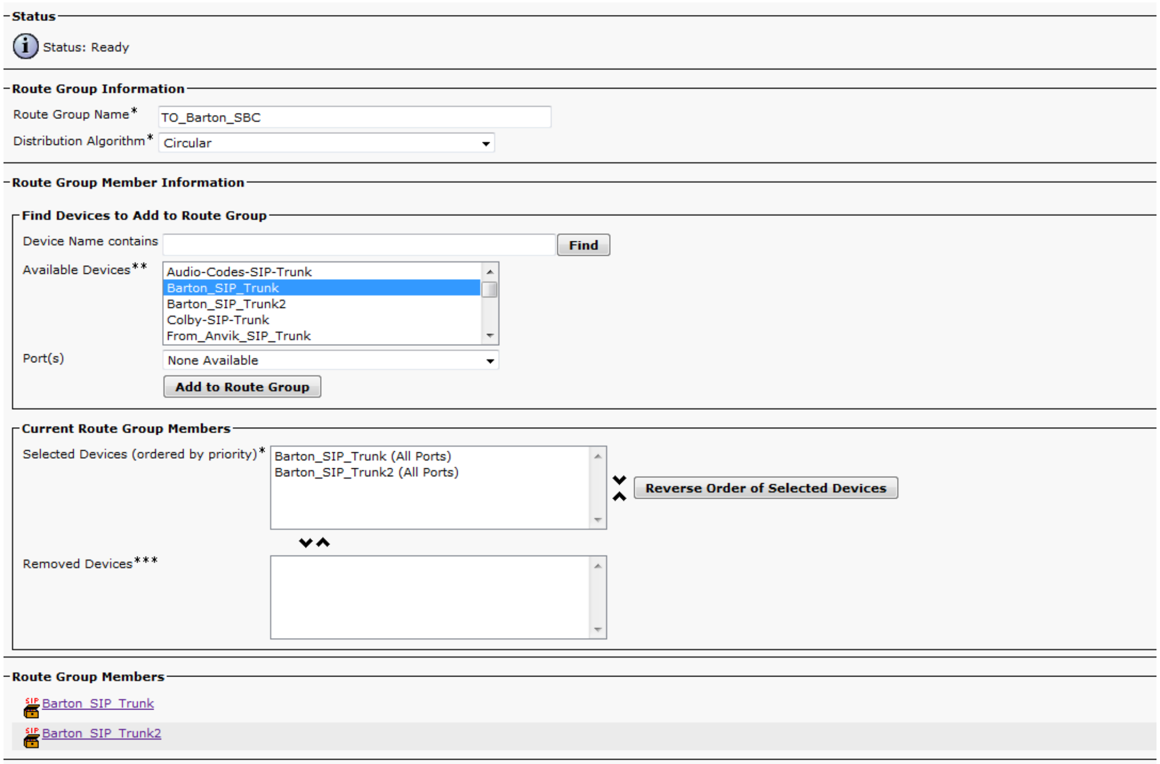
MLPP Domain	< None >
Confidential Access Mode	< None >
Confidential Access Level	< None >



4.Route Group

Select **Call Routing > Route/Hunt > Route Group > Add New**

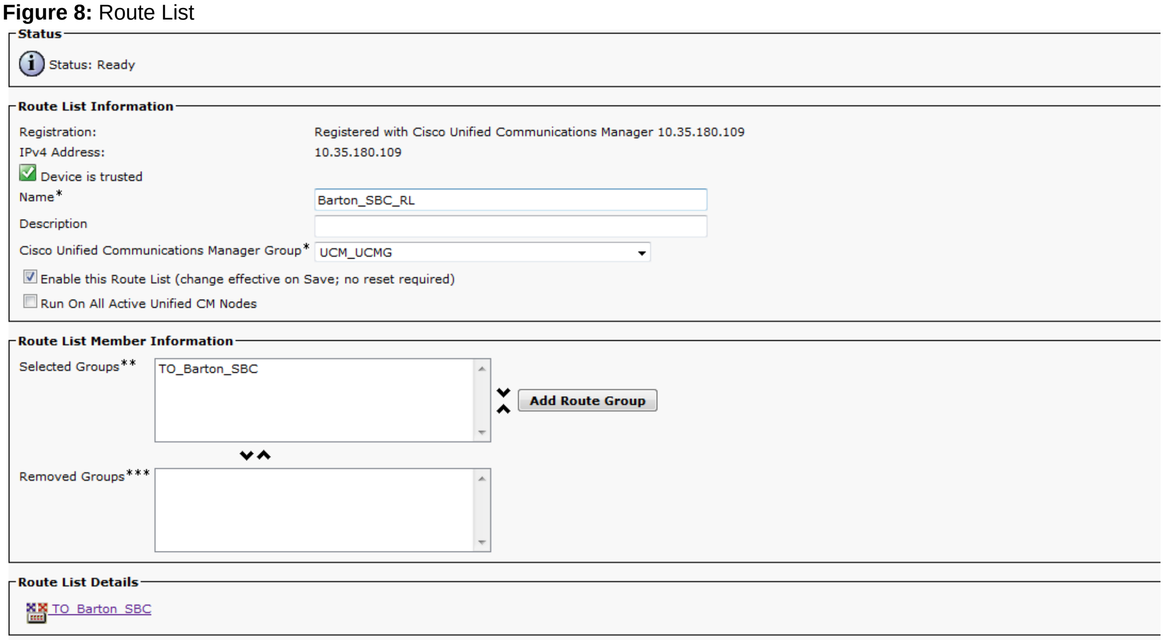
Figure 7: Route Group



5.Route List

Select **Call Routing > Route/Hunt > Route List > Add New**

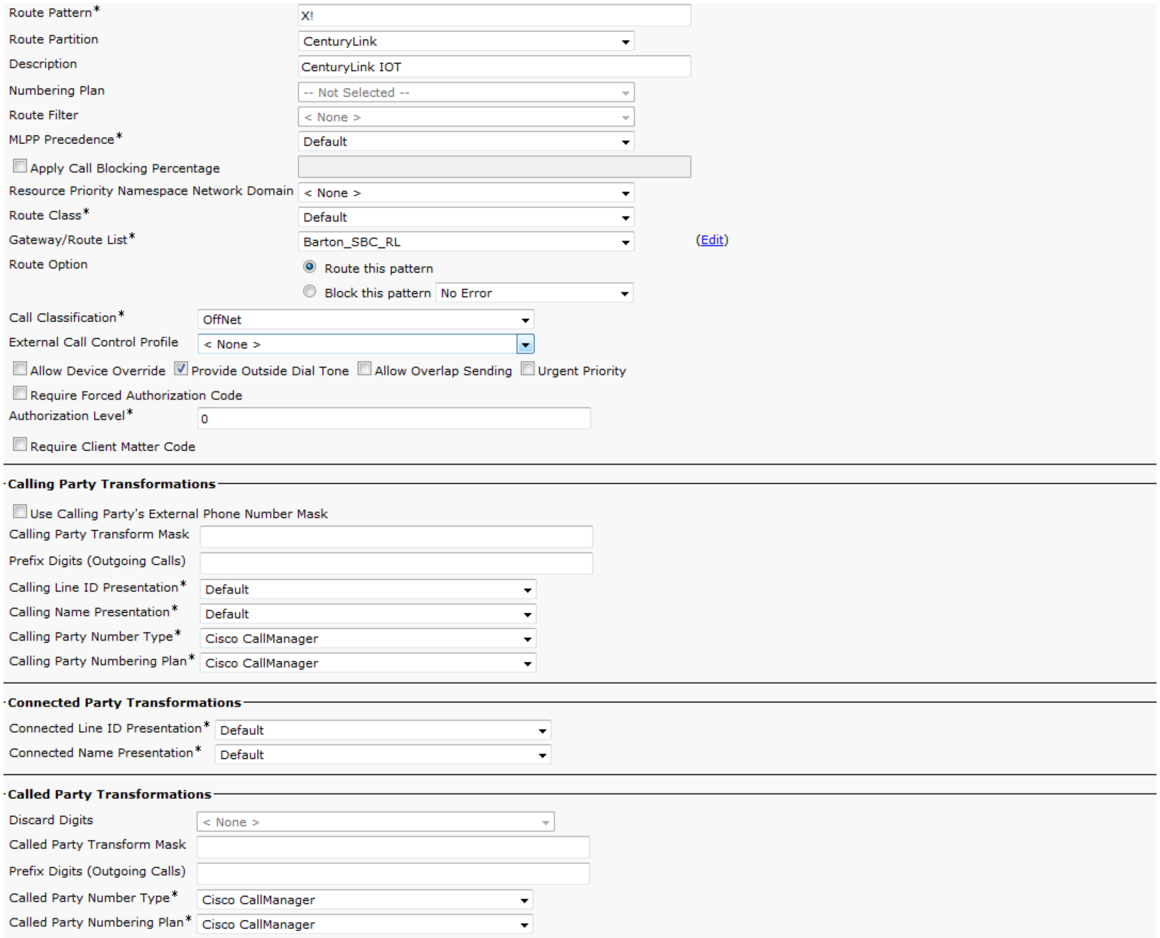
Figure 8: Route List



6.Route Pattern

Select **Call Routing > Route/Hunt > Route Pattern > Add New**

Figure 9: Route Pattern



Sonus SBC 5XX0 Series Configuration

Complete Configuration

```
configure

#UDP Port Range for RTP (media)
set system media mediaPortRange baseUdpPort 1024 maxUdpPort 65148
commit

#DSP Resources
set system mediaProfile compression 90 tone 10
commit

#CUCM codecs
set profiles media codecEntry G711u_CUCM codec g711 packetSize 20 law ULaw
set profiles media codecEntry G711u_CUCM dtmf relay rfc2833 removeDigits enable
set profiles media codecEntry G711u_CUCM fax failureHandling continue toneTreatment none
set profiles media codecEntry G711u_CUCM modem failureHandling continue toneTreatment none
commit

#CenturyLink codecs
set profiles media codecEntry G729_CenturyLink codec g729a packetSize 20 preferredRtpPayloadType 128
set profiles media codecEntry G711u_CenturyLink codec g711 packetSize 20 law ULaw
set profiles media codecEntry G711u_CenturyLink dtmf relay rfc2833 removeDigits enable
set profiles media codecEntry G729_CenturyLink dtmf relay rfc2833 removeDigits enable
```

```
set profiles media codecEntry G711u_CenturyLink fax failureHandling continue toneTreatment faxRelay
set profiles media codecEntry G729_CenturyLink fax failureHandling continue toneTreatment faxRelay
set profiles media codecEntry G729_CenturyLink modem failureHandling continue toneTreatment
applyFaxTreatment
set profiles media codecEntry G711u_CenturyLink modem failureHandling continue toneTreatment
applyFaxTreatment
commit

#Tenor fax codecs
set profiles media codecEntry G729_Tenor_T38 codec g729a packetSize 20 preferredRtpPayloadType 128
set profiles media codecEntry G711u_Tenor_T38 codec g711 packetSize 20 law ULaw
set profiles media codecEntry G711u_Tenor_T38 dtmf relay rfc2833 removeDigits enable
set profiles media codecEntry G729_Tenor_T38 dtmf relay rfc2833 removeDigits enable
set profiles media codecEntry G711u_Tenor_T38 fax failureHandling continue toneTreatment faxRelay
set profiles media codecEntry G729_Tenor_T38 fax failureHandling continue toneTreatment faxRelay
set profiles media codecEntry G729_Tenor_T38 modem failureHandling continue toneTreatment
applyFaxTreatment
set profiles media codecEntry G711u_Tenor_T38 modem failureHandling continue toneTreatment
applyFaxTreatment
commit

#Internal Side Configuration
#IP Interface Group
set addressContext default ipInterfaceGroup Private ipInterface Private-pkt0 ceName BARTONCE portName
pkt0 ipAddress 10.35.177.238 prefix 26 mode outOfService state disabled
set addressContext default ipInterfaceGroup Private ipInterface Private-pkt0 mode inService state
enabled
commit

#IP Static Route
set addressContext default staticRoute 0.0.0.0 0 10.35.177.193 Private Private-pkt0 preference 100
commit

#SBC Configuration for CUCM10.5 Trunk
#Packet Service Profile (PSP)
set profiles media packetServiceProfile CUCM10_PSP packetToPacketControl codecsAllowedForTranscoding
thisLeg g711u,g729 otherLeg g711u,g729
set profiles media packetServiceProfile CUCM10_PSP packetToPacketControl transcode conditional
set profiles media packetServiceProfile CUCM10_PSP rtcpOptions rtcp enable terminationForPassthrough
enable
set profiles media packetServiceProfile CUCM10_PSP silenceInsertionDescriptor g711SidRtpPayloadType 13
heartbeat enable
commit

#IP Signaling profiles(IPSP)
set profiles signaling ipSignalingProfile CUCM10_IPSP ipProtocolType sipOnly
set profiles signaling ipSignalingProfile CUCM10_IPSP commonIpAttributes flags disableMediaLockDown
enable includeReasonHeader enable
set profiles signaling ipSignalingProfile CUCM10_IPSP commonIpAttributes flags
minimizeRelayingOfMediaChangesFromOtherCallLegAll enable publishIPInHoldSDP enable sendPtimeInSdp
enable
set profiles signaling ipSignalingProfile CUCM10_IPSP commonIpAttributes flags sendRtcpPortInSdp enable
set profiles signaling ipSignalingProfile CUCM10_IPSP commonIpAttributes relayFlags notify enable
set profiles signaling ipSignalingProfile CUCM10_IPSP commonIpAttributes transparencyFlags  mwiBody
enable  unknownBody enable unknownHeader enable
set profiles signaling ipSignalingProfile CUCM10_IPSP ingressIpAttributes flags
sendSdpIn200OkIf18xReliable enable sendSdpInSubsequent18x enable
set profiles signaling ipSignalingProfile CUCM10_IPSP egressIpAttributes flags disable2806Compliance
enable
set profiles signaling ipSignalingProfile CUCM10_IPSP egressIpAttributes flags
disableOptionalRegisterParameters enable
set profiles signaling ipSignalingProfile CUCM10_IPSP egressIpAttributes transport type1 tcp type2 none
type3 none type4 none
commit

#Signaling profiles(SP)
set profiles signaling signalingProfile CUCM10_SP egress egressFlags addPrefix011ForInternationalCalls
enable convertNumbersToE164Format enable
commit

#Transparency profile(TP)
set profiles services transparencyProfile CUCM10_TP sipMessageBody all excludedMethods register invite
```

```
subscribe info publish message refer options update
set profiles services transparencyProfile CUCM10_TP state enabled
commit

#E.164 Profile
set profiles signaling E164Profile CenturyLink_E164  sonusE164ProfCharStar allow sonusE164ProfCharHash
allow sonusE164ProfCharHyphen allow
commit

#zone
set addressContext default zone ZONE-INT-VoIP id 3
commit

set addressContext default zone ZONE-INT-CUCM id 4
commit

#SIP signaling port
set addressContext default zone ZONE-INT-VoIP sipSigPort 25 ipInterfaceGroupName Private ipAddressV4
10.35.177.239 portNumber 5060 mode outOfService state disabled siprec disabled
transportProtocolsAllowed sip-tcp
set addressContext default zone ZONE-INT-VoIP sipSigPort 25 mode inService state enabled
commit
set addressContext default zone ZONE-INT-CUCM sipSigPort 1 ipInterfaceGroupName Private ipAddressV4
10.35.177.239 portNumber 5070 mode outOfService state disabled siprec disabled
transportProtocolsAllowed sip-tcp
set addressContext default zone ZONE-INT-CUCM sipSigPort 1 mode inService state enabled
commit

#IP Peer
set addressContext default zone ZONE-INT-VoIP ipPeer CUCM10_SIP_SERVER policy sip fqdnPort 0
set addressContext default zone ZONE-INT-VoIP ipPeer CUCM10_SIP_SERVER ipAddress 10.35.180.109 ipPort
5060
set addressContext default zone ZONE-INT-VoIP ipPeer CUCM10_SIP_SERVER authentication
incInternalCredentials enabled
set addressContext default zone ZONE-INT-VoIP ipPeer CUCM10_SIP_SERVER authentication
intChallengeResponse enabled
set addressContext default zone ZONE-INT-VoIP ipPeer CUCM10_SIP_SERVER surrogateRegistration userPart
4696806537 authUserName xxxxxxx-4696806537 regAuthPassword xxxx state enabled sendCredentials
challengeForAnyMessageAndInDialogRequests hostPart voip.centurylink.com
commit

set addressContext default zone ZONE-INT-CUCM ipPeer CUCM10_SIP_SERVER2 policy sip fqdnPort 0
set addressContext default zone ZONE-INT-CUCM ipPeer CUCM10_SIP_SERVER2 ipAddress 10.35.180.109 ipPort
5070 defaultForIp false
set addressContext default zone ZONE-INT-CUCM ipPeer CUCM10_SIP_SERVER2 authentication
incInternalCredentials enabled
set addressContext default zone ZONE-INT-CUCM ipPeer CUCM10_SIP_SERVER2 authentication
intChallengeResponse enabled
set addressContext default zone ZONE-INT-CUCM ipPeer CUCM10_SIP_SERVER2 surrogateRegistration userPart
4696806538 authUserName xxxxxx-4696806538 regAuthPassword xxxx state enabled sendCredentials
challengeForAnyMessageAndInDialogRequests hostPart voip.centurylink.com
commit

#SIP trunk group
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup CUCM10_TG media mediaIpInterfaceGroupName
Private sourceAddressFiltering disabled
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup CUCM10_TG signaling messageManipulation
inputAdapterProfile CenturyLink_HashIn includeAppHdrs disabled
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup CUCM10_TG signaling rel100Support enabled
acceptHistoryInfo enabled
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup CUCM10_TG ingressIpPrefix 10.35.180.109 32
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup CUCM10_TG policy callRouting
elementRoutingPriority ERP_CenturyLink
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup CUCM10_TG policy digitParameterHandling
numberingPlan CenturyLink_NANP
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup CUCM10_TG policy media packetServiceProfile
CUCM10_PSP
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup CUCM10_TG policy services classOfService
DEFAULT_IP
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup CUCM10_TG policy signaling
ipSignalingProfile CUCM10_IPSP signalingProfile CUCM10_SP
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup CUCM10_TG services transparencyProfile
```

```
CUCM10_TP
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup CUCM10_TG signaling E164Profiles
e164LocalProfile CenturyLink_E164 e164GlobalProfile CenturyLink_E164
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup CUCM10_TG state enabled mode inService
commit

set addressContext default zone ZONE-INT-CUCM sipTrunkGroup CUCM10_TG2 media mediaIpInterfaceGroupName
Private sourceAddressFiltering disabled
set addressContext default zone ZONE-INT-CUCM sipTrunkGroup CUCM10_TG2 signaling messageManipulation
inputAdapterProfile CenturyLink_HashIn includeAppHdrs disabled
set addressContext default zone ZONE-INT-CUCM sipTrunkGroup CUCM10_TG2 signaling rel100Support enabled
acceptHistoryInfo enabled
set addressContext default zone ZONE-INT-CUCM sipTrunkGroup CUCM10_TG2 ingressIpPrefix 10.35.180.109 32
set addressContext default zone ZONE-INT-CUCM sipTrunkGroup CUCM10_TG2 policy callRouting
elementRoutingPriority ERP_CenturyLink
set addressContext default zone ZONE-INT-CUCM sipTrunkGroup CUCM10_TG2 policy digitParameterHandling
numberingPlan CenturyLink_NANP
set addressContext default zone ZONE-INT-CUCM sipTrunkGroup CUCM10_TG2 policy media
packetServiceProfile CUCM10_PSP
set addressContext default zone ZONE-INT-CUCM sipTrunkGroup CUCM10_TG2 policy services classOfService
DEFAULT_IP
set addressContext default zone ZONE-INT-CUCM sipTrunkGroup CUCM10_TG2 policy signaling
ipSignalingProfile CUCM10_IPSP signalingProfile CUCM10_SP
set addressContext default zone ZONE-INT-CUCM sipTrunkGroup CUCM10_TG2 services transparencyProfile
CUCM10_TP
set addressContext default zone ZONE-INT-CUCM sipTrunkGroup CUCM10_TG2 signaling E164Profiles
e164LocalProfile CenturyLink_E164 e164GlobalProfile CenturyLink_E164
set addressContext default zone ZONE-INT-CUCM sipTrunkGroup CUCM10_TG2 state enabled mode inService
commit


#SBC Configuration for Fax Trunk
#Packet Service Profile
set profiles media packetServiceProfile TENOR_FAX_PSP codec codecEntry1 G711u_Tenor_T38 codecEntry2
G729_Tenor_T38
set profiles media packetServiceProfile TENOR_FAX_PSP packetToPacketControl codecsAllowedForTranscoding
thisLeg g711u,g729,t38 otherLeg g711u,g729,t38
set profiles media packetServiceProfile TENOR_FAX_PSP packetToPacketControl transcode only
set profiles media packetServiceProfile TENOR_FAX_PSP rtcpOptions rtcp enable
commit

#IP signaling profile
set profiles signaling ipSignalingProfile TENOR_FAX_IPSP ipProtocolType sipOnly
set profiles signaling ipSignalingProfile TENOR_FAX_IPSP commonIpAttributes flags disableMediaLockDown
enable includeReasonHeader enable
set profiles signaling ipSignalingProfile TENOR_FAX_IPSP commonIpAttributes flags
minimizeRelayingOfMediaChangesFromOtherCallLegAll enable publishIPInHoldSDP enable sendPtimeInSdp
enable
set profiles signaling ipSignalingProfile TENOR_FAX_IPSP commonIpAttributes flags sendRtcpPortInSdp
enable
set profiles signaling ipSignalingProfile TENOR_FAX_IPSP commonIpAttributes relayFlags notify enable
set profiles signaling ipSignalingProfile TENOR_FAX_IPSP commonIpAttributes transparencyFlags  mwiBody
enable  unknownBody enable unknownHeader enable
set profiles signaling ipSignalingProfile TENOR_FAX_IPSP ingressIpAttributes flags
sendSdpIn200OkIf18xReliable enable sendSdpInSubsequent18x enable
set profiles signaling ipSignalingProfile TENOR_FAX_IPSP egressIpAttributes flags disable2806Compliance
enable
set profiles signaling ipSignalingProfile TENOR_FAX_IPSP egressIpAttributes transport type1 udp type2
none type3 none type4 none
commit

#Signaling profiles
set profiles signaling signalingProfile TENOR_FAX_SP egress egressFlags
addPrefix011ForInternationalCalls enable convertNumbersToE164Format enable
commit

#IP peer
set addressContext default zone ZONE-INT-VoIP ipPeer TENOR_SIP_SERVER policy sip fqdnPort 0
set addressContext default zone ZONE-INT-VoIP ipPeer TENOR_SIP_SERVER ipAddress 10.35.137.43 ipPort
5084 defaultForIp false
set addressContext default zone ZONE-INT-VoIP ipPeer TENOR_SIP_SERVER authentication
incInternalCredentials enabled
```

```
set addressContext default zone ZONE-INT-VoIP ipPeer TENOR_SIP_SERVER authentication
intChallengeResponse enabled
set addressContext default zone ZONE-INT-VoIP ipPeer TENOR_SIP_SERVER surrogateRegistration userPart
4696806537 authUserName xxxxxxx-4696806537 regAuthPassword xxxx state disabled sendCredentials
challengeForAnyMessageAndInDialogRequests hostPart voip.centurylink.com
commit

#SIP trunk group
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup TENOR_TG_FAX media
mediaIpInterfaceGroupName Private
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup TENOR_TG_FAX signaling messageManipulation
inputAdapterProfile CenturyLink_HashIn includeAppHdrs disabled
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup TENOR_TG_FAX signaling rel100Support
enabled acceptHistoryInfo enabled
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup TENOR_TG_FAX ingressIpPrefix 10.35.137.43
32
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup TENOR_TG_FAX policy callRouting
elementRoutingPriority ERP_CenturyLink
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup TENOR_TG_FAX policy digitParameterHandling
numberingPlan CenturyLink_NANP
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup TENOR_TG_FAX policy media
packetServiceProfile TENOR_FAX_PSP
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup TENOR_TG_FAX policy services classOfService
DEFAULT_IP
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup TENOR_TG_FAX policy signaling
ipSignalingProfile TENOR_FAX_IPSP signalingProfile TENOR_FAX_SP
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup TENOR_TG_FAX services transparencyProfile
CenturyLink
set addressContext default zone ZONE-INT-VoIP sipTrunkGroup TENOR_TG_FAX state enabled mode inService
commit


#External Side SBC Configuration

#IP Interface Group
set addressContext default zone ZONE-EXT-ACCESS sipSigPort 10 ipInterfaceGroupName Public ipAddressV4
216.110.2.227 portNumber 5060 mode outOfService state disabled transportProtocolsAllowed sip-udp
sip-tcp
set addressContext default zone ZONE-EXT-ACCESS sipSigPort 10 mode inService state enabled
commit

set addressContext default ipInterfaceGroup Public ipInterface Public-pkt2 ceName BARTONCE portName
pkt2 ipAddress 216.110.2.226 prefix 28 mode outOfService state disabled
set addressContext default ipInterfaceGroup Public ipInterface Public-pkt2 mode inService state enabled
commit

#IP static route
set addressContext default staticRoute 216.206.64.0 24 216.110.2.225 Public Public-pkt2 preference 100
commit

#SBC Configuration for CenturyLink SIP Trunk

#Packet Service Profile (PSP)
set profiles media packetServiceProfile CenturyLink_PSP packetToPacketControl
codecsAllowedForTranscoding thisLeg g711u,g729 otherLeg g711u,g729
set profiles media packetServiceProfile CenturyLink_PSP packetToPacketControl transcode conditional
set profiles media packetServiceProfile CenturyLink_PSP rtcpOptions rtcp enable
terminationForPassthrough enable
set profiles media packetServiceProfile CenturyLink_PSP silenceInsertionDescriptor
g711SidRtpPayloadType 13 heartbeat enable
commit

#IP Signaling profiles(IPSP)
set profiles signaling ipSignalingProfile CenturyLink_IPSP ipProtocolType sipOnly
set profiles signaling ipSignalingProfile CenturyLink_IPSP commonIpAttributes flags
disableMediaLockDown enable includeReasonHeader enable
set profiles signaling ipSignalingProfile CenturyLink_IPSP commonIpAttributes flags
minimizeRelayingOfMediaChangesFromOtherCallLegAll enable publishIPInHoldSDP enable sendPtimeInSdp
enable
set profiles signaling ipSignalingProfile CenturyLink_IPSP commonIpAttributes flags sendRtcpPortInSdp
enable
set profiles signaling ipSignalingProfile CenturyLink_IPSP commonIpAttributes relayFlags notify enable
```

```
set profiles signaling ipSignalingProfile CenturyLink_IPSP commonIpAttributes transparencyFlags
mwiBody enable  unknownBody enable unknownHeader enable
set profiles signaling ipSignalingProfile CenturyLink_IPSP ingressIpAttributes flags
sendSdpIn200OkIf18xReliable enable sendSdpInSubsequent18x enable
set profiles signaling ipSignalingProfile CenturyLink_IPSP egressIpAttributes flags
disable2806Compliance enable
set profiles signaling ipSignalingProfile CenturyLink_IPSP egressIpAttributes transport type1 udp type2
none type3 none type4 none
set profiles signaling ipSignalingProfile CenturyLink_IPSP egressIpAttributes flags
disableOptionalRegisterParameters enable
commit

#Signaling profiles(SP)
set profiles signaling signalingProfile CenturyLink_SP egress egressFlags
addPrefix011ForInternationalCalls enable convertNumbersToE164Format enable
commit

#Transparency profile(TP)
set profiles services transparencyProfile CenturyLink_TP sipMessageBody all excludedMethods register
invite subscribe info publish message refer options update
set profiles services transparencyProfile CenturyLink_TP state enabled
commit

#DM/PM rule
set profiles digitParameterHandling dmPmCriteria CenturyLink_HashPrefix criteriaType digit digitType
calledNumber parameterPresenceCheck exists
set profiles digitParameterHandling dmPmCriteria CenturyLink_HashPrefix digitCriteria digitMatch value
startDigitPosition 0 numberOfDigits 3 matchValue 999
set profiles digitParameterHandling dmPmCriteria CenturyLink_HashPrefix digitCriteria digitMatch
operation equals
set profiles digitParameterHandling dmPmCriteria CenturyLink_HashPrefix digitCriteria egressFlag value
send operation ignore
set profiles digitParameterHandling dmPmCriteria CenturyLink_HashPrefix digitCriteria natureOfAddress
value 950 operation ignore
set profiles digitParameterHandling dmPmCriteria CenturyLink_HashPrefix digitCriteria numberLength
value 5 operation equals
commit

set profiles digitParameterHandling dmPmRule CenturyLink_Hash_Add subRule 0 criteria
CenturyLink_HashPrefix ruleType digit
set profiles digitParameterHandling dmPmRule CenturyLink_Hash_Add subRule 0 digitManipulation
digitStringManipulation replacement type constant digitString calledNumber startDigitPosition 0
numberOfDigits 1 value #
set profiles digitParameterHandling dmPmRule CenturyLink_Hash_Add subRule 0 digitManipulation
digitStringManipulation startDigitPosition 0 numberOfDigits 3 action none
set profiles digitParameterHandling dmPmRule CenturyLink_Hash_Add subRule 0 digitManipulation
numberParameterManipulation natureOfAddress none numberingPlanIndicator none numberLength 3
presentation none screening none includeInEgress none
set profiles digitParameterHandling dmPmRule CenturyLink_Hash_Add subRule 0 digitManipulation
numberType calledNumber
commit

#Zone
set addressContext default zone ZONE-EXT-ACCESS id 20
commit

#SIP signaling port
set addressContext default zone ZONE-EXT-ACCESS sipSigPort 10 ipInterfaceGroupName Public ipAddressV4
216.110.2.227 portNumber 5060 mode outOfService state disabled transportProtocolsAllowed sip-udp
set addressContext default zone ZONE-EXT-ACCESS sipSigPort 10 mode inService state enabled
commit

#IP peer
set addressContext default zone ZONE-EXT-ACCESS ipPeer CenturyLink_IPP policy sip fqdnPort 0
set addressContext default zone ZONE-EXT-ACCESS ipPeer CenturyLink_IPP ipAddress 216.206.64.91 ipPort
5100 defaultForIp false
set addressContext default zone ZONE-EXT-ACCESS ipPeer CenturyLink_IPP2 policy sip fqdnPort 0
set addressContext default zone ZONE-EXT-ACCESS ipPeer CenturyLink_IPP2 ipAddress 216.206.66.91 ipPort
5100 defaultForIp false
commit

#SIP trunk group
```

```
set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG media
mediaIpInterfaceGroupName Public
set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG signaling
messageManipulation outputAdapterProfile CenturyLink_SMM includeAppHdrs disabled
set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG signaling rel100Support
enabled acceptHistoryInfo enabled
set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG ingressIpPrefix
216.206.64.91 32
set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG policy callRouting
elementRoutingPriority ERP_CenturyLink
set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG policy
digitParameterHandling numberingPlan CenturyLink_NANP egressDmPmRule CenturyLink_Hash_Add
set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG policy media
packetServiceProfile CenturyLink_PSP
set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG policy services
classOfService DEFAULT_IP
set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG policy signaling
ipSignalingProfile CenturyLink_IPSP signalingProfile CenturyLink_SP
set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG services
transparencyProfile CenturyLink_TP
set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG state enabled mode
inService
commit

set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG2 media
mediaIpInterfaceGroupName Public
set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG2 signaling
messageManipulation outputAdapterProfile CenturyLink_SMM2 includeAppHdrs disabled
set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG2 signaling rel100Support
enabled acceptHistoryInfo enabled
set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG2 ingressIpPrefix
216.206.66.91 32
set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG2 policy callRouting
elementRoutingPriority ERP_CenturyLink
set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG2 policy
digitParameterHandling numberingPlan CenturyLink_NANP egressDmPmRule CenturyLink_Hash_Add
set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG2 policy media
packetServiceProfile CenturyLink_PSP
set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG2 policy services
classOfService DEFAULT_IP
set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG2 policy signaling
ipSignalingProfile CenturyLink_IPSP signalingProfile CenturyLink_SP
set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG2 services
transparencyProfile CenturyLink_TP
set addressContext default zone ZONE-EXT-ACCESS sipTrunkGroup CenturyLink_TG2 state enabled mode
inService
commit

#Global Call Routing Configuration
#Element Routing Priority
set profiles callRouting elementRoutingPriority ERP_CenturyLink
set profiles callRouting elementRoutingPriority ERP_CenturyLink entry nationalType 2 entityType none
set profiles callRouting elementRoutingPriority ERP_CenturyLink entry userName 2 entityType none
set profiles callRouting elementRoutingPriority ERP_CenturyLink entry internationalType 2 entityType
none
set profiles callRouting elementRoutingPriority ERP_CenturyLink entry nationalType 1 entityType
trunkGroup
set profiles callRouting elementRoutingPriority ERP_CenturyLink entry internationalType 1 entityType
trunkGroup
set profiles callRouting elementRoutingPriority ERP_CenturyLink entry userName 1 entityType trunkGroup
commit

#CUCM10 Routing
set global callRouting routingLabel TO_CUCM10_TG routingLabelRoute 0 trunkGroup CUCM10_TG ipPeer
CUCM10_SIP_SERVER proportion 100 cost 100 inService inService testing normal
set global callRouting routingLabel TO_CUCM10_TG overflowNOA none overflowNPI none
routePrioritizationType sequence action routes numRoutesPerCall 10
set global callRouting routingLabel TO_CUCM10_TG2 routingLabelRoute 0 trunkGroup CUCM10_TG2 ipPeer
CUCM10_SIP_SERVER2 proportion 100 cost 100 inService inService testing normal
set global callRouting routingLabel TO_CUCM10_TG2 overflowNOA none overflowNPI none
routePrioritizationType sequence action routes numRoutesPerCall 10
set global callRouting route trunkGroup CUCM10_TG BARTONCE standard Sonus_NULL Sonus_NULL all all ALL
```

```
none Sonus_NULL routingLabel TO_CenturyLink
set global callRouting route trunkGroup CUCM10_TG2 BARTONCE standard Sonus_NULL Sonus_NULL all all ALL
none Sonus_NULL routingLabel TO_CenturyLink2
commit

#CenturyLink routing
set global callRouting routingLabel TO_CenturyLink routingLabelRoute 0 trunkGroup CenturyLink_TG ipPeer
CenturyLink_IPP proportion 100 cost 100 inService inService testing normal
set global callRouting routingLabel TO_CenturyLink overflowNOA none overflowNPI none
routePrioritizationType proportionAllocation action routes numRoutesPerCall 10
set global callRouting routingLabel TO_CenturyLink2 routingLabelRoute 0 trunkGroup CenturyLink_TG2
ipPeer CenturyLink_IPP2 proportion 100 cost 100 inService inService testing normal
set global callRouting routingLabel TO_CenturyLink2 overflowNOA none overflowNPI none
routePrioritizationType proportionAllocation action routes numRoutesPerCall 10
set global callRouting route trunkGroup CenturyLink_TG BARTONCE standard Sonus_NULL Sonus_NULL all all
ALL none Sonus_NULL routingLabel TO_CUCM10_TG
set global callRouting route trunkGroup CenturyLink_TG2 BARTONCE standard Sonus_NULL Sonus_NULL all all
ALL none Sonus_NULL routingLabel TO_CUCM10_TG2
commit


#SMM
set profiles signaling sipAdaptorProfile CenturyLink_HashIn rule 1 criterion 1 type message
set profiles signaling sipAdaptorProfile CenturyLink_HashIn rule 1 criterion 1 message messageTypes all
set profiles signaling sipAdaptorProfile CenturyLink_HashIn rule 1 applyMatchHeader one
set profiles signaling sipAdaptorProfile CenturyLink_HashIn rule 1 criterion 2 type header
set profiles signaling sipAdaptorProfile CenturyLink_HashIn rule 1 criterion 2 header name To condition
exist
set profiles signaling sipAdaptorProfile CenturyLink_HashIn rule 1 action 1 type header operation
regsub
set profiles signaling sipAdaptorProfile CenturyLink_HashIn rule 1 action 1 to type header value To
set profiles signaling sipAdaptorProfile CenturyLink_HashIn rule 1 action 1 regexp string %23
matchInstance all
set profiles signaling sipAdaptorProfile CenturyLink_HashIn rule 1 action 1 from type value value 999
set profiles signaling sipAdaptorProfile CenturyLink_HashIn rule 2 criterion 2 type header
set profiles signaling sipAdaptorProfile CenturyLink_HashIn rule 2 criterion 2 type header
set profiles signaling sipAdaptorProfile CenturyLink_HashIn rule 2 criterion 2 header name Request-Line
condition exist
set profiles signaling sipAdaptorProfile CenturyLink_HashIn rule 2 criterion 1 type message
set profiles signaling sipAdaptorProfile CenturyLink_HashIn rule 2 criterion 1 message messageTypes all
set profiles signaling sipAdaptorProfile CenturyLink_HashIn rule 2 applyMatchHeader one
set profiles signaling sipAdaptorProfile CenturyLink_HashIn rule 2 action 1 type header operation
regsub
set profiles signaling sipAdaptorProfile CenturyLink_HashIn rule 2 action 1 to type header value
Request-Line
set profiles signaling sipAdaptorProfile CenturyLink_HashIn rule 2 action 1 regexp string %23
set profiles signaling sipAdaptorProfile CenturyLink_HashIn rule 2 action 1 from type value value 999
set profiles signaling sipAdaptorProfile CenturyLink_HashIn state enable
commit

set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 1 criterion 1 type message
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 1 criterion 1 message messageTypes all
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 1 criterion 2 type header
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 1 criterion 2 header name
P-Asserted-Identity condition exist
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 1 applyMatchHeader one
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 1 action 1 type header operation delete
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 1 action 1 to type header value
P-Asserted-Identity
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 2 criterion 1 type message
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 2 criterion 1 message messageTypes all
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 2 criterion 2 type header
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 2 criterion 2 header name
P-Asserted-Identity condition absent
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 2 applyMatchHeader one
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 2 action 1 type header operation add
headerPosition last
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 2 action 1 to type header value
P-Asserted-Identity
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 2 action 1 from type value value
<sip:4696806537@voip.centurylink.com>
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 3 criterion 1 type message
```

```
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 3 criterion 1 message messageTypes
request methodTypes register
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 3 criterion 2 type header
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 3 criterion 2 header name Request-Line
condition exist
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 3 criterion 3 type token
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 3 criterion 3 token condition exist
tokenType urihostname
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 3 applyMatchHeader one
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 3 action 1 type token operation regsub
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 3 action 1 to type token tokenValue
urihostname
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 3 action 1 regexp string
voip.centurylink.com matchInstance all
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 3 action 1 from type value value
216.206.64.91
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 4 criterion 1 type message
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 4 criterion 1 message messageTypes all
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 4 criterion 2 type header
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 4 criterion 2 header name From condition
exist
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 4 applyMatchHeader one
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 4 action 1 type header operation regsub
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 4 action 1 to type header value From
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 4 action 1 regexp string
anonymous@216.110.2.227 matchInstance all
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 4 action 1 from type value value
anonymous@anonymous.invalid
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 4 action 2 type header operation regsub
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 4 action 2 to type header value From
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 4 action 2 regexp string
Restricted@216.110.2.227 matchInstance all
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 4 action 2 from type value value
anonymous@anonymous.invalid
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 5 criterion 1 type message
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 5 criterion 1 message messageTypes all
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 5 criterion 2 type header
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 5 criterion 2 header name Contact
condition exist
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 5 action 1 type header operation regsub
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 5 action 1 to type header value Contact
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 5 action 1 regexp string
sip:anonymous@216.110.2.227:5060 matchInstance all
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 5 action 1 from type value value
sip:anonymous@anonymous.invalid:5060
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 5 action 2 type header operation regsub
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 5 action 2 to type header value Contact
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 5 action 2 regexp string
sip:Restricted@216.110.2.227:5060 matchInstance all
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 5 action 2 from type value value
sip:anonymous@anonymous.invalid:5060
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 6 criterion 1 type message
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 6 criterion 1 message messageTypes all
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 6 criterion 2 type header
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 6 criterion 2 header name From condition
exist
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 6 applyMatchHeader one
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 6 action 1 type header operation regsub
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 6 action 1 to type header value From
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 6 action 1 regexp string 216.110.2.227
matchInstance all
set profiles signaling sipAdaptorProfile CenturyLink_SMM rule 6 action 1 from type value value
voip.centurylink.com
set profiles signaling sipAdaptorProfile CenturyLink_SMM state enabled
commit
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 1 criterion 1 type message
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 1 criterion 1 message messageTypes all
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 1 criterion 2 type header
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 1 criterion 2 header name
P-Asserted-Identity condition exist
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 1 applyMatchHeader one
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 1 action 1 type header operation delete
```

```
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 1 action 1 to type header value
P-Asserted-Identity
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 2 criterion 1 type message
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 2 criterion 1 message messageTypes all
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 2 criterion 2 type header
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 2 criterion 2 header name
P-Asserted-Identity condition absent
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 2 applyMatchHeader one
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 2 action 1 type header operation add
headerPosition last
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 2 action 1 to type header value
P-Asserted-Identity
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 2 action 1 from type value value
<sip:4696806538@voip.centurylink.com>
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 3 criterion 1 type message
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 3 criterion 1 message messageTypes
request methodTypes register
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 3 criterion 2 type header
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 3 criterion 2 header name Request-Line
condition exist
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 3 criterion 3 type token
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 3 criterion 3 token condition exist
tokenType urihostname
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 3 applyMatchHeader one
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 3 action 1 type token operation regsub
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 3 action 1 to type token tokenValue
urihostname
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 3 action 1 regexp string
voip.centurylink.com matchInstance all
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 3 action 1 from type value value
216.206.66.91
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 4 criterion 1 type message
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 4 criterion 1 message messageTypes all
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 4 criterion 2 type header
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 4 criterion 2 header name From condition
exist
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 4 applyMatchHeader one
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 4 action 1 type header operation regsub
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 4 action 1 to type header value From
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 4 action 1 regexp string
anonymous@216.110.2.227 matchInstance all
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 4 action 1 from type value value
anonymous@anonymous.invalid
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 4 action 2 type header operation regsub
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 4 action 2 to type header value From
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 4 action 2 regexp string
Restricted@216.110.2.227 matchInstance all
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 4 action 2 from type value value
anonymous@anonymous.invalid
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 5 criterion 1 type message
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 5 criterion 1 message messageTypes all
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 5 criterion 2 type header
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 5 criterion 2 header name Contact
condition exist
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 5 action 1 type header operation regsub
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 5 action 1 to type header value Contact
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 5 action 1 regexp string
sip:anonymous@216.110.2.227:5060 matchInstance all
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 5 action 1 from type value value
sip:anonymous@anonymous.invalid:5060
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 5 action 2 type header operation regsub
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 5 action 2 to type header value Contact
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 5 action 2 regexp string
sip:Restricted@216.110.2.227:5060 matchInstance all
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 5 action 2 from type value value
sip:anonymous@anonymous.invalid:5060
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 6 criterion 1 type message
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 6 criterion 1 message messageTypes all
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 6 criterion 2 type header
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 6 criterion 2 header name From condition
exist
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 6 applyMatchHeader one
```

```
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 6 action 1 type header operation regsub
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 6 action 1 to type header value From
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 6 action 1 regexp string 216.110.2.227
matchInstance all
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 rule 6 action 1 from type value value
```

```
voip.centurylink.com
set profiles signaling sipAdaptorProfile CenturyLink_SMM2 state enabled
commit
```

Test Results

S.No	Title	Description	Test Setup	Result	Comment
g729-001	Anonymous Call Rejection Activate	PBX User dials *77 PSTN Calls PBX User with Caller ID Block Should receive an announcement	*77 is Dialed PBX and leaves PBX Phones gets an announcement Calling Party blocks caller ID Calling party makes a call to PBX User Calling Party receives an announcement when PBX user is dialed	Passed	
g729-002	Anonymous Call Rejection Deactivate	PBX User dials *87 PSTN Calls PBX User with Caller ID block Call Should Complete	*87 is dialed PBX User receives and announcement PSTN calls PBX User PSTN Phone receives ringback PBX Phone gets ringing PBX Phone get Caller ID PBX Phone answer the Call 2 way audio is received PBX Phone releases Calls PSTN receives a Bye	Passed	

g729-003	Anonymous Call PBX-BW	PBX sends anonymous call to BW BW delivers the calls Private or unknown or anonymous to PSTN	PBX is configured to send a call to BW as anonymous with TN as PSTN BW delivers the call to PSTN as Private or Anonymous PSTN phone shows the call as Private or Anonymous Call is answered by PSTN PBX user hangs up the call	Passed	
g729-004	Alien TNs	A call PBX call originate where the from TN that is not part of the customer trunk group. As long as the pilot number is identified in outgoing call by PAI, the BroadWorks will accept and route the call.	After Alien TN is set up on a Trunk in CenturyLink Network PBX User Places a Call to PSTN PBX User receives ringback PSTN receives ringing PSTN receives caller id of the Alien TN PSTN answers the call 2 way audio is received PBX Phone releases Calls PSTN receives a Bye	Passed	

g729-005	Barge In	Create a Pick Up Group with 2 PBX Users PSTN Calls PBX User 1 PBX User 2 dials *33 +PBX User Ext PSTN, User 1, and User 2 should be conf	PSTN calls PBX User 1 PSTN Phone receives ringback PBX Phone gets ringing PBX Phone get Caller ID PBX Phone answer the Call 2 way audio is received PBX User 2 Dials *33 + PBX User 1 Extension PSTN, PBX User 1, and PBX User 2 are conferenced together 2 Way Audio is heard by all Legs PBX User 1 drops from Call 2 way Audio is heard by PSTN and PBX User 2 PSTN drops call PBX User 2 receives a Bye	Passed	
g729-006	Barge In Exempt	In the Portal Enable Barge In Exempt Create a Pick Up Group with 2 PBX Users PSTN Calls PBX User 1 PBX User 2 dials *33 +PBX User Ext User 2 Should not be conf	Barge in Exempt is set on PBX user 1 PSTN calls PBX User 1 PSTN Phone receives ringback PBX Phone gets ringing PBX Phone get Caller ID PBX Phone answer the Call 2 way audio is received PBX User 2 Dials *33 + PBX User 1 Extension PBX user 2 is not allowed to barge in PSTN drops the call PBX User 1 receives a Bye	Passed	

g729-007	PSTN to BWA	PSTN calls BWA Number Enter Calling Number (2nd Phone Location) Enter Called Number (PSTN) PSTN should Ring with Caller ID of 2nd Phone Answer Call	BroadWorks Anywhere is set up in Portal PSTN 1 Calls BWA Number Announcement is received Enter calling Number (2nd Phone created in BWA) Announcement received Enter Called Number (PSTN 2) PSTN 1 receives ringback PSTN 2 receives ringing PSTN 2 receives caller ID of 2nd Phone (Not of PSTN 1) PSTN 2 Answers Call 2 way audio is received PSTN 2 releases Calls PSTN receives a Bye	Not Run	Anywhere service is not activated for test account
g729-008	PSTN to PBX user with BWA	PSTN Calls User with BWA PBX User and 2nd Location should Ring Answer phone for 2nd location	BroadWorks Anywhere is set up in Portal PSTN 1 Calls BWA Number PSTN 1 receives ringback Both PBX User and 2nd Phone Location Number gets ringing Both PBX User and 2nd Phone Location Number gets Caller ID of PSTN Call is answered on Location 2 PBX User no longer gets ringing (cancel) 2 way Audio Location 2 releases call PSTN receives a Bye	Not Run	Anywhere service is not activated for test account

g729-009	Call Forwarding Always Activate	PBX User dials *72 Enter the CFA Destination TN PSTN calls PBX User with CFA	PBX User 1 Dials *72 Announcement is heard PBX User enter PBX User 2 TN Announcement is heard PBX Receives a Bye PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 does not ring PBX User 2 gets ringing PBX user 2 receives Caller ID (PSTN Originator Caller) PBX User answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed	
g729-010	Call Forwarding Always Interrogate	PBX User with CFA dials *21* Announcement received	PBX User 1 Dials *21* Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	
g729-011	Call Forwarding Always Deactivate	PBX User with CFA dials *73 PSTN Calls PBX User	PBX User 1 Dials *73 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	

g729-012	Call Forwarding Always to Voicemail Activate	PBX User Dials *21 PSTN Dials PBX User with CFA Verify Call goes to Voicemail	PBX User 1 Dials *21 Announcement is received When announcement completes PBX User receives a Bye PSTN User Calls PBX User 1 Call should go directly to voicemail Announcement is Heard Leave voicemail After leaving voicemail PSTN should receive a Bye PBX User 1 should receive and MWI	Passed	
g729-013	Call Forwarding Always to Voicemail Deactivate	PBX User with CFA dial #21 PSTN dials PBX User verify Phone rings	PBX User 1 Dials #21 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	
g729-014	PSTN call is CFB to PSTN with ID Restricted	PBX configured to send CFB to BW for identified Station. BW is configured with CFB to PSTN2. PSTN 1 Calls PBX with Caller ID Restricted PSTN 1 hears ring back PBX send 486 Busy to BW BW forwards the call to PSTN2 PSTN 2 hears ringing PSTN 2 Caller ID displays Private/Anonymous PSTN 2 Answers the call. Two way voice path is established between PSTN 1 and PSTN 2 PSTN 2 hangs up	PSTN2 should receive Private/Anonymous as CLID	Passed	
g729-015	PSTN with Privacy call to PBX is CFA to PSTN	PBX User is configured with CFA to PSTN 2 PSTN 1 Calls PBX with Caller ID Restricted PSTN 1 hears ring back PBX sends a new call to BW with PSTN 2 Number, From as Anonymous and PAI set to Pilot Number BW forwards the call to PSTN2 PSTN 2 hears ringing PSTN 2 Caller ID displays Pilot Number PSTN 2 Answers the call. Two way voice path is established between PSTN 1 and PSTN 2 PSTN 2 hangs up	Pilot Number should be shown as CLID on PSTN2	Passed	

g729-016	Call Forwarding Busy Activate	PBX User dials *90 Enter the CFB Destination TN PSTN calls PBX User with CFB	PBX User 1 Dials *90 Announcement is heard PBX User enter PBX User 2 TN Announcement is heard PBX Receives a Bye Busy PBX User 1 PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 does not ring PBX User 2 gets ringing PBX user 2 receives Caller ID (PSTN Originator Caller) PBX User answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed	
g729-017	Call Forwarding Busy Interrogate	PBX User with CFB dials *67* Announcement received	PBX User 1 Dials *67* Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	
g729-018	Call Forwarding Busy Deactivate	PBX User with CFB dials *91 PSTN Calls PBX User	PBX User 1 Dials *91 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	

g729-019	Call Forwarding Busy to Voicemail Activate	PBX User Dials *40 PSTN Dials PBX User with CFB Verify Call goes to Voicemail	PBX User 1 Dials *40 Announcement is received When announcement completes PBX User receives a Bye Busy PBX User 1 PSTN User Calls PBX User 1 Call should go directly to voicemail Announcement is Heard Leave voicemail After leaving voicemail PSTN should receive a Bye PBX User 1 should receive and MWI	Passed	
g729-020	Call Forwarding Busy to Voicemail Deactivate	PBX User with CFB dial #40 PSTN dials PBX User verify Phone rings	PBX User 1 Dials #40 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	

g729-021	Call Forwarding No Answer Activate	PBX User dials *92 Enter the CFNA Destination TN PSTN calls PBX User with CFNA	PBX User 1 Dials *92 Announcement is heard PBX User enters PBX User 2 TN Announcement is heard PBX Receives a Bye PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX User 1 receives Caller ID After timer is RNA is received PBX User 2 gets ringing PBX user 2 receives Caller ID PBX User 2 answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed	

g729-022	Call Forwarding No Answer- RNA Timer	PBX User dials *610 Enter 1 # PSTN calls PBX User with CFNA Verify Call is forwarded	PBX User 1 Dials *610 Announcement is Heard PBX User enter 1 for amount of Rings After announcement completes PBX User 1 receives a Bye PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX User 1 receives Caller ID After timer is RNA is received PBX User 2 gets ringing PBX user 2 receives Caller ID PBX User 2 answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed	Minimum is 0 or 2 rings which can be entered
g729-023	Call Forwarding No Answer Interrogate	PBX User with CFNA dials *61* Announcement received	PBX User 1 Dials *61* Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	
g729-024	Call Forwarding No Answer Deactivate	PBX User with CFNA dials *93 PSTN Calls PBX User	PBX User 1 Dials *93 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	

g729-025	Call Forwarding No Answer to Voicemail Activate	PBX User Dials *41 PSTN Dials PBX User with CFNA Verify Call goes to Voicemail	PBX User 1 Dials *41 Announcement is received When announcement completes PBX User receives a Bye Busy PBX User 1 PSTN User Calls PBX User 1 Call should go to voicemail after RNA timer is reached Announcement is Heard Leave voicemail After leaving voicemail PSTN should receive a Bye PBX User 1 should receive and MWI	Passed	
g729-026	Call Forwarding No Answer to Voicemail Deactivate	PBX User with CFNA dial #41 PSTN dials PBX User verify Phone rings	PBX User 1 Dials #41 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	

g729-027	Call Forwarding Not Reachable Activate	PBX User dials *94 Enter the CFNR Destination TN Unregister Pilot TNs PSTN calls PBX User with CFNR Verify Call is forwarded Register Pilot TNs	PBX User 1 Dials *94 Announcement is heard PBX User enter PBX User 2 TN Announcement is heard PBX Receives a Bye Unplug SBC Lan Cable PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 does not ring PSTN User 2 gets ringing PSTN user 2 receives Caller ID (PSTN Originator Caller) PSTN User answers call 2 way Audio PSTN User 1 releases call PSTN User 2 receives a Bye	Passed	
g729-028	Call Forwarding Not Reachable Interrogate	PBX User with CFNR dials *63* Announcement received	PBX User 1 Dials *63* Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	
g729-029	Call Forwarding Not Reachable Deactivate	PBX User with CFNR dials *95 PSTN Calls PBX User	PBX User 1 Dials *95 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	

g729-030	Call Forwarding Selective Activate	Log into Portal and set up Call forward selective User with a PSTN Number PBX User with CFS enters #76 PSTN User calls PBX User with CFS Call should be call forwarded	Log into Portal and set up Call forward selective User with a PSTN Number PBX User with CFS dials #76 Announcement received PBX User receives a Bye From a Selected PSTN Dial PBX User 1 PBX User should not Ring Call should be call forwarded to the CFS Destination PSTN receives Ringback Destination receives Ringing Destination receives Caller ID (Originator PSTN) Destination answers call 2 way Audio PSTN ends the call Destination receives a Bye	Passed	
g729-031	Call Forwarding Selective Deactivate	PBX User with CFS enters #77 PSTN User calls PBX User Call should not be forwarded	PBX User 1 Dials #77 Announcement is heard PBX Receives a Bye PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed	

g729-032	Call Return by PBX User	PBX User dials *69	PSTN 1 Calls PBX User 1 PSTN 1 receives ringback PBX User 1 receives ringing PBX User 1 receives caller ID PBX User 1 answers call 2 way Audio PSTN 1 ends the call PBX User 1 receives a Bye PBX User 1 Dials *69 PBX User receives Ringback PSTN 1 receives Ringing PSTN receives Caller ID PSTN answers 2 way Audio PSTN releases call PBX User 1 receives a Bye	Passed	
g729-033	Consultative Transfer with SIP REFER	PBX User Calls PSTN PBX User transfers PSTN to PSTN2 PBX User has Audio with PSTNs PSTN 1 has MOH PBX User Transfers Call PSTN and PSTN2 now have audio		Not Run	CUCM 10.5 does not support outbound SIP Transfer with Refer method
g729-034	Unattended Transfer with SIP REFER	PBX User Calls PSTN PBX User transfers PSTN to PSTN2 During Ringback PBX User transfers PSTN 1 has MOH PSTN2 answers call PSTN and PSTN2 now have audio		Not Run	CUCM 10.5 does not support outbound SIP Transfer with Refer method

g729-035	Consultative Transfer	PBX User Calls PSTN PBX User transfers PSTN to PBX User 2 PBX User 1 has Audio with PBX User 2 PSTN 1 has MOH PBX User Transfers Call PSTN and PBX 2 now have audio	PBX User 1 Calls PSTN PBX User receives Ringback PSTN 1 receives Ringing PSTN 1 receives Caller ID PSTN 1 answers 2 way Audio PBX User transfers call to PBX User 2 PSTN User gets MOH PBX User 1 gets Dial tone PBX User 1 dials PBX User 2 Extension PBX User 1 receives Ringback PBX User 2 receives Ringing PBX User 2 receives Caller ID of PBX User 1 PBX User 2 answers the Call 2 way Audio PBX User 1 transfers the call MOH Ends PSTN 1 and PBX User 2 are now connected 2 Way Audio PSTN 1 Ends the call PBX User 2 receives the Bye	Passed	

g729-036	Unattended Transfer	PBX User Calls PSTN PBX User transfers PSTN to PBX User 2 During Ringback PBX User transfers PSTN 1 has MOH PBX User 2 answers call PSTN and PBX User 2 now have audio	PBX User 1 Calls PSTN PBX User receives Ringback PSTN 1 receives Ringing PSTN 1 receives Caller ID PSTN 1 answers 2 way Audio PBX User transfers call to PBX User 2 PSTN User gets MOH PBX User 1 gets Dial tone PBX User 1 dials PBX User 2 Extension PBX User 1 receives Ringback PBX User 2 receives Ringing PBX User 2 receives Caller ID of PSTN 1 PBX User 1 release call PBX User 2 answers the Call MOH Ends 2 way Audio PSTN 1 release the call PBX User 2 receives the Bye	Passed	

g729-037	Call Waiting Persistent Activate	PBX User dials *43 PSTN Calls PBX User PSTN 2 Calls PBX User Verify Call Waiting Tone	PBX User 1 Dials *43 Announcement is heard PBX Receives a Bye PSTN User 1 Calls PBX User 1 PSTN User 1 receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN User 2 Calls PBX User 1 PSTN User 2 receives ringback PBX User 1 receives caller ID PBX User 1 hear Call Waiting Tone PBX User Places PSTN User 1 on Hold PSTN User 1 hears MOH PBX User 1 answers Call from PSTN 2 2 way Audio Verify PBX User 1 can swap between to callers While on PBX User 1 and PSTN User 1 PSTN 1 releases Call PBX User 1 receives a Bye Call 2 should still be up with PSTN 2 hearing MOH	Passed	

g729-038	Call Waiting Persistent Deactivate	PBX User Dials #43 PSTN Calls PBX User PSTN 2 Calls PBX User Call 2 should go to voicemail	PBX User 1 Dials #43 Announcement is heard PBX User 1 Receives a Bye after Announcement is completed PSTN User Calls PBX User 1 PSTN User 1 Calls PBX User 1 PSTN User 1 receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User 1 answers call 2 way Audio PSTN User 1 releases the call PBX User 1 receives a Bye	Passed	Call Forwarding Busy to Voicemail is activated to send PSTN User 2 to voicemail
g729-039	Customer Originated Trace	PSTN Calls PBX User PBX User Answers the Call PBX User Hangs up call PBX User enters *57 Verify announcement	PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye PBX User 1 Dial *57 Announcement received Announcement Completes PBX User receives a Bye	Passed	
g729-040	Enhanced Call Logs	Log into portal and verify Call logs	Log into the portal for PBX User 1 On main screen verify calls Logs are displayed Missed Received Placed	Passed	

g729-041	Last Number Redial	PBX User dials *66 The last number dialed should be called	PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN releases call PBX User 1 receives a Bye PBX User 1 Dial *66 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN releases call PBX User 1 receives a Bye	Passed	
g729-042	MOH	Verify MOH for conference, transfer, and hold	PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PBX User 1 Places call on Hold PSTN receives MOH PBX User retrieves call from Hold 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed	

g729-043	Remote Office - Like CFA	Provision Remote office for a SIP Trunk user on the BroadWorks portal to use PSTN number A. Place a call from a PSTN number B to the SIP Trunk user's DID and verify that it is forwarded to PSTN number A (the destination configured in BroadWorks).	Log into the portal for PBX User 1 Set up remote Office to PSTN Number 1 PSTN User 2 Calls PBX User 1 PSTN 2 receives ringback PSTN User 1 gets ringing with PSTN 2 Caller ID and Diversion header for PBX User1 PSTN User 1 answers call 2 way Audio PSTN 1 releases call	Passed	
g729-044	Remote Office - Quick Call	Provision Remote office for a SIP Trunk user 1 on the BroadWorks portal to use PSTN number A. On the BW portal, Manage Users, select Configure Features of User 1, under Quick Call, add PSTN B number and click on the Call Button. PSTN A should Start Ringing with PBX User 1 Caller ID.	Log into the portal for PBX User 1 Set up remote Office to PSTN Number 1 Initiate a Quack Call to PSTN 2 on the portal PSTN User 1 gets ringing with PBX User 1 Caller ID PSTN user 1 answers the call. Now PSTN2 should start ringing with PBX User1 as Caller ID. PSTN 1 might hear ringback based on how long PSTN 2 rings. PSTN 2 answers the call 2 way Audio PSTN 1 releases call	Passed	

g729-045	Remote Office - Click to Call	Provision Remote office for a SIP Trunk user 1 on the BroadWorks portal to use PSTN number A. On the BW portal, Manage Users, select Configure Features of User 1, under Call Logs, select either incoming/outgoing/missed calls and Click on a Call under Phone Number Click To call column. PSTN A should Start Ringing with PBX User 1 Caller ID.	Log into the portal for PBX User 1 Set up remote Office to PSTN Number 1 Review call logs and identify a call log that needs to be called via Click to Call. Click on the identified call log under Click to Call PSTN User 1 gets ringing with PBX User 1 Caller ID PSTN user 1 answers the call. Now PSTN2 should start ringing with PBX User1 as Caller ID. PSTN 1 might hear ringback based on how long PSTN 2 rings. PSTN 2 answers the call 2 way Audio PSTN 1 releases call	Passed	
g729-046	Selective Call Acceptance	Provision selective call acceptance in the BroadWorks portal. Place a call from an accepted TN to the SIP Trunk User. Verify that the call completes normally. Place a call from a TN that is not on the accept list and verify that the call does not reach the SBC.	Log into the portal for PBX User 1 Set up Selected Call Acceptance to PSTN Number 1 PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN releases Call PBX User 1 receives a Bye	Passed	PSTN number 2144326686 was configured

g729-047	Selective Call Rejection	Provision selective call rejection in the BroadWorks portal. Place a call from a TN not on the reject list to the SIP Trunk User. Verify that the call completes normally. Place a call from a TN that is on the reject list and verify that the call does not reach the SBC.	Log into the portal for PBX User 1 Set up Selected Call rejection to PSTN Number 1 PSTN Calls PBX User 1 Verify PSTN gets an announcement PSTN receives a Bye	Passed	PSTN number 2144326687 was configured
g729-048	Sequential Ring	Provision sequential ring in the BroadWorks portal. Place a call to the SIP trunk user. Verify that the numbers in the sequential ring list are dialed in order.	Log into the Portal for PBX User 1 Set up Sequential Ring with PBX User 2 and PBX User 3 PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID After RNA Timer PBX User 1 receives a Cancel PBX User 2 gets ringing PBX user 2 receives Caller ID After RNA Timer PBX User 1 receives a Cancel PBX User 3 gets ringing PBX user 3 receives Caller ID PBX User 3 answers call 2 way Audio PSTN releases Call PBX User 3 receives a Bye	Passed	4696806539, 6540,6541

g729-049	Simultaneous Ring	Provision Simultaneous ring in the BroadWorks portal. Place a call to the SIP trunk user. Verify that the numbers in the Simultaneous ring list are dialed at once.	Log into the Portal for PBX User 1 Set up Simultaneous Ring with PBX User 2 and PBX User 3 PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User 2 gets ringing PBX user 2 receives Caller ID PBX User 3 gets ringing PBX user 3 receives Caller ID PBX User 3 Answers Call PBX User 1 and 2 receive a Cancel 2 way Audio PSTN releases Call PSTN User 3 receives a Bye	Passed	
g729-050	Third Party MWI Control NOTIFY	Provision Third Party MWI in the BroadWorks portal. Provision the CT Voice Mail system to notify BroadWorks of unread messages in the user's voice mail box. Confirm that the NOTIFY is sent to BroadWorks and that the NOTIFY is sent to the PBX.	PSTN User Calls PBX User 1 Call should go to voicemail after RNA timer is reached Announcement is Heard Leave voicemail After leaving voicemail PSTN should receive a Bye PBX User 1 should receive and MWI PBX User 1 dials *86 Log into Mailbox Listen To Voicemail Delete Voicemail Verify MWI is gone PBX User 1 ends the Call	Passed	Call Forwarding No Answer to Voicemail is activated to send PSTN user 1 to voicemail after timer

g729-051	Voice Mail Consultation	Provision Voice Mail n the BroadWorks or NYMPH portal. Provision the PBX to forward calls to an external voice mail system as the user's call coverage. Confirm the PBX user's capability to retrieve voice mail from the external Voice Mail system.	PSTN User Calls PBX User 1 Call should go to voicemail after RNA timer is reached Announcement is Heard Leave voicemail After leaving voicemail PSTN should receive a Bye PBX User 1 should receive and MWI PBX User 1 dials *86 Log into Mailbox Listen To Voicemail Delete Voicemail Verify MWI is gone PBX User 1 ends the Call	Passed	

g729-052	PBX Initiate Conference	PBX User Calls PSTN PBX User Conferences PBX User 2	PBX User 1 Calls PSTN PBX User receives Ringback PSTN 1 receives Ringing PSTN 1 receives Caller ID PSTN 1 answers 2 way Audio PBX User conferences call to PBX User 2 PSTN User gets MOH PBX User 1 gets Dial tone PBX User 1 dials PBX User 2 Extension PBX User 1 receives Ringback PBX User 2 receives Ringing PBX User 2 receives Caller ID of PBX User 1 PBX User 2 answers the Call 2 way Audio PBX User 1 conferences the call MOH Ends PSTN 1, PBX User 1 and PBX User 2 are now connected 2 Way Audio PBX User 1 Ends the call PBX User 2 and PSTN receives the Bye	Passed	

g729-053	PSTN Initiate Conference	PBX User calls PSTN PSTN conferences PBX User2	PBX User 1 Calls PSTN PBX User receives Ringback PSTN 1 receives Ringing PSTN 1 receives Caller ID PSTN 1 answers 2 way Audio PSTN User 1 conferences call to PBX User 2 PBX User 1 gets MOH PSTN User 1 gets Dial tone PSTN User 1 dials PBX User 2 Extension PSTN User 1 receives Ringback PBX User 2 receives Ringing PBX User 2 receives Caller ID of PSTN User 1 PBX User 2 answers the Call 2 way Audio PSTN User 1 conferences the call MOH Ends PSTN 1, PBX User 1 and PBX User 2 are now connected 2 Way Audio PSTN User 1 Ends the call PBX User 1 and PBX User 2 Still Have Audio PBX User 1 End the Call PBX User 2 receives a Bye	Passed	

g729-054	Huntgroup Seq Ring	PSTN Calls Huntgroup Seq ring Answer call on 2nd Member	Log into Admin Portal Create Huntgroup with 3 members PSTN Calls Huntgroup PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID After RNA Timer is reached PBX User 2 gets ringing PBX user 2 receives Caller ID PBX User 2 Answers the call 2 way Audio PSTN ends the call PBX User 2 receives a Bye	Blocked	Hunt group is not purchased

g729-055	Huntgroup Seq Ring RNA to Voicemail	PSTN calls Huntgroup Seq ring RNA to Voicemail	Log into Admin Portal Create Huntgroup with 3 members PSTN Calls Huntgroup PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID After RNA Timer is reached PBX User 2 gets ringing PBX user 2 receives Caller ID After RNA Timer is reached PBX User 3 gets ringing PBX user 3 receives Caller ID Call should go to voicemail after RNA timer is reached Announcement is Heard Leave voicemail After leaving voicemail PSTN should receive a Bye PBX User 1 should receive and MWI PBX User 1 dials *86 Enter *# Log into HuntGroup Mailbox Listen To Voicemail Delete Voicemail PBX User 1 ends the Call	Not Run	Hunt group is not activated for test account

g729-056	Huntgroup Sim Ring	PSTN calls Huntgroup Sim ring 3 members Answer Call	Log into Admin Portal Create Huntgroup with 3 members with Sequential Ring PSTN Calls Huntgroup PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User 2 gets ringing PBX user 2 receives Caller ID PBX user 3 receives Caller ID PBX User 3 Answers the call PBX User 3 Answers the Call 2 way Audio PSTN ends the call PBX User 2 receives a Bye	Not Run	Hunt group is not activated for test account
g729-057	PBX to PBX	PBX User Calls PBX User2 Same Trunk Verify RTP is dropped to SBC	PBX User 1 Calls PBX User 2 PBX User 1 receives ringback PBX User 2 gets ringing PBX user 2 receives Caller ID PBX User 2 answers call 2 way Audio RTP is on SBC/PBX PBX User 1 End the call PBX User 2 receives a Bye	Passed	
g729-058	PSTN to PBX	PSTN to PBX User	PSTN User 1 Calls PBX User 1 PSTN User 1 receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN User Ends The Call PBX User 1 receives a Bye	Passed	

g729-059	PBX to PSTN	PBX User to PSTN	PBX User 1 Calls PSTN User 1 PBX User 1 receives ringback PSTN User 1 gets ringing PSTN user 1 receives Caller ID PSTN User answers call 2 way Audio PSTN User Ends The Call PBX User 1 receives a Bye	Passed	
g729-060	PBX to PBX Different PBX (diff realm)	PBX User to PBX User Different PBX (diff realm)	PBX User 1 Calls PBX User 2 Diff Realm PBX User 1 receives ringback PBX User 2 gets ringing PBX user 2 receives Caller ID PBX User 2 answers call 2 way Audio RTP PBX User 1 End the call PBX User 2 receives a Bye	Passed	
g729-061	PSTN to PBX	PSTN to PBX User Fax Call	PSTN User 1 Fax Calls PBX User 1 Fax PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User 1 Fax answers call Fax is received PBX User Ends The Call PSTN User 1 receives a Bye	Passed	

g729-062	PBX to PSTN	PBX User to PSTN Fax Call	PBX User 1 Fax Calls PSTN User 1 Fax PSTN User 1 gets ringing PSTN user 1 receives Caller ID PSTN User 1 Fax answers call Fax is received PSTN User Ends The Call PBX User 1 receives a Bye	Passed	
g729-063	PSTN to PBX -T38	PSTN to PBX User Fax Call	PSTN User 1 Fax Calls PBX User 1 Fax PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User 1 Fax answers call Fax is received PBX User Ends The Call PSTN User 1 receives a Bye	Passed	
g729-064	PBX to PSTN -T38	PBX User to PSTN Fax Call	PBX User 1 Fax Calls PSTN User 1 Fax PSTN User 1 gets ringing PSTN user 1 receives Caller ID PSTN User 1 Fax answers call Fax is received PSTN User Ends The Call PBX User 1 receives a Bye	Passed	
g729-065	PBX to PSTN - Packet Marking for SIG packets	PBX to PSTN Call to verify that signaling packets are marked with DSCP = 24 (0x18)	All outgoing SIP Signaling packets are marked with DSCP=24	Passed	Same as g729-062
g729-066	PBX to PSTN - Packet Marking for RTP packets	PBX to PSTN Call to verify that rtp packets are marked with DSCP = 40 (0x28)	All outgoing SIP RTP packets are marked with DSCP=40	Passed	Same as g729-062

g729-067	PBX to PSTN - Directory assistance	PBX User Calls PBX 411 and speaks with directory assistant	PBX User 1 dials 411 Call is delivered to Directory Assistant for enquiry Once the user hears an announcement or speaks with an operator, PBX user hangs up the call	Passed	
g729-068	PBX to PSTN - Toll Free	PBX User Calls 800.366.8201 to test toll free numbers	PBX User 1 dials 800.366.8201 (CTL Support) Call is delivered to CenturyLink Support Once the user hears an announcement or speaks with an operator, PBX user hangs up the call	Passed	
g729-069	PBX to PSTN - 911	PBX User Calls 911 to get emergency support	PBX User 1 dials xxx-xxx-xxxx (CTL Rep) Call is delivered to CenturyLink Rep PBX User makes conferences 911 operator PBX User, CTL rep and 911 operator are conferenced ????	Passed	Same routing as for g729-067
g729-070	PBX to PSTN - International	PBX User Calls international number	International Call is successfully established and torn down.	Passed	
g711-001	Anonymous Call Rejection Activate	PBX User dials *77 PSTN Calls PBX User with Caller ID Block Should receive an announcement	*77 is Dialed PBX and leaves PBX Phones gets an announcement Calling Party blocks caller ID Calling party makes a call to PBX User Calling Party receives an announcement when PBX user is dialed	Passed	

g711-002	Anonymous Call Rejection Deactivate	PBX User dials *87 PSTN Calls PBX User with Caller ID block Call Should Complete	*87 is dialed PBX User receives and announcement PSTN calls PBX User PSTN Phone receives ringback PBX Phone gets ringing PBX Phone get Caller ID PBX Phone answer the Call 2 way audio is received PBX Phone releases Calls PSTN receives a Bye	Passed	
g711-003	Anonymous Call PBX-BW	PBX sends anonymous call to BW BW delivers the calls Private or unknown or anonymous to PSTN	PBX is configured to send a call to BW as anonymous with TN as PSTN BW delivers the call to PSTN as Private or Anonymous PSTN phone shows the call as Private or Anonymous Call is answered by PSTN PBX user hangs up the call	Passed	
g711-004	Alien TNs	A call PBX call originate where the from TN that is not part of the customer trunk group. As long as the pilot number is identified in outgoing call by PAI, the BroadWorks will accept and route the call.	After Alien TN is set up on a Trunk in CenturyLink Network PBX User Places a Call to PSTN PBX User receives ringback PSTN receives ringing PSTN receives caller id of the Alien TN PSTN answers the call 2 way audio is received PBX Phone releases Calls PSTN receives a Bye	Passed	

g711-005	Barge In	Create a Pick Up Group with 2 PBX Users PSTN Calls PBX User 1 PBX User 2 dials *33 +PBX User Ext PSTN, User 1, and User 2 should be conf	PSTN calls PBX User 1 PSTN Phone receives ringback PBX Phone gets ringing PBX Phone get Caller ID PBX Phone answer the Call 2 way audio is received PBX User 2 Dials *33 + PBX User 1 Extension PSTN, PBX User 1, and PBX User 2 are conferenced together 2 Way Audio is heard by all Legs PBX User 1 drops from Call 2 way Audio is heard by PSTN and PBX User 2 PSTN drops call PBX User 2 receives a Bye	Passed	
g711-006	Barge In Exempt	In the Portal Enable Barge In Exempt Create a Pick Up Group with 2 PBX Users PSTN Calls PBX User 1 PBX User 2 dials *33 +PBX User Ext User 2 Should not be conf	Barge in Exempt is set on PBX user 1 PSTN calls PBX User 1 PSTN Phone receives ringback PBX Phone gets ringing PBX Phone get Caller ID PBX Phone answer the Call 2 way audio is received PBX User 2 Dials *33 + PBX User 1 Extension PBX user 2 is not allowed to barge in PSTN drops the call PBX User 1 receives a Bye	Passed	

g711-007	PSTN to BWA	PSTN calls BWA Number Enter Calling Number (2nd Phone Location) Enter Called Number (PSTN) PSTN should Ring with Caller ID of 2nd Phone Answer Call	BroadWorks Anywhere is set up in Portal PSTN 1 Calls BWA Number Announcement is received Enter calling Number (2nd Phone created in BWA) Announcement received Enter Called Number (PSTN 2) PSTN 1 receives ringback PSTN 2 receives ringing PSTN 2 receives caller ID of 2nd Phone (Not of PSTN 1) PSTN 2 Answers Call 2 way audio is received PSTN 2 releases Calls PSTN receives a Bye	Not Run	Anywhere service is not activated for test account
g711-008	PSTN to PBX user with BWA	PSTN Calls User with BWA PBX User and 2nd Location should Ring Answer phone for 2nd location	BroadWorks Anywhere is set up in Portal PSTN 1 Calls BWA Number PSTN 1 receives ringback Both PBX User and 2nd Phone Location Number gets ringing Both PBX User and 2nd Phone Location Number gets Caller ID of PSTN Call is answered on Location 2 PBX User no longer gets ringing (cancel) 2 way Audio Location 2 releases call PSTN receives a Bye	Not Run	Anywhere service is not activated for test account

g711-009	Call Forwarding Always Activate	PBX User dials *72 Enter the CFA Destination TN PSTN calls PBX User with CFA	PBX User 1 Dials *72 Announcement is heard PBX User enter PBX User 2 TN Announcement is heard PBX Receives a Bye PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 does not ring PBX User 2 gets ringing PBX user 2 receives Caller ID (PSTN Originator Caller) PBX User answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed	
g711-010	Call Forwarding Always Interrogate	PBX User with CFA dials *21* Announcement received	PBX User 1 Dials *21* Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	
g711-011	Call Forwarding Always Deactivate	PBX User with CFA dials *73 PSTN Calls PBX User	PBX User 1 Dials *73 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	

g711-012	Call Forwarding Always to Voicemail Activate	PBX User Dials *21 PSTN Dials PBX User with CFA Verify Call goes to Voicemail	PBX User 1 Dials *21 Announcement is received When announcement completes PBX User receives a Bye PSTN User Calls PBX User 1 Call should go directly to voicemail Announcement is Heard Leave voicemail After leaving voicemail PSTN should receive a Bye PBX User 1 should receive and MWI	Passed	
g711-013	Call Forwarding Always to Voicemail Deactivate	PBX User with CFA dial #21 PSTN dials PBX User verify Phone rings	PBX User 1 Dials #21 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	
g711-014	PSTN with Privacy call to PBX is CFA to PSTN	PBX User is configured with CFA to PSTN 2 PSTN 1 Calls PBX with Caller ID Restricted PSTN 1 hears ring back PBX sends a new call to BW with PSTN 2 Number, From as Anonymous and PAI set to Pilot Number BW forwards the call to PSTN2 PSTN 2 hears ringing PSTN 2 Caller ID displays Pilot Number PSTN 2 Answers the call. Two way voice path is established between PSTN 1 and PSTN 2 PSTN 2 hangs up	Pilot Number should be shown as CLID on PSTN2	Passed	
g711-015	PSTN call is CFB to PSTN with ID Restricted	PBX configured to send CFB to BW for identified Station. BW is configured with CFB to PSTN2. PSTN 1 Calls PBX with Caller ID Restricted PSTN 1 hears ring back PBX send 486 Busy to BW BW forwards the call to PSTN2 PSTN 2 hears ringing PSTN 2 Caller ID displays Private/Anonymous PSTN 2 Answers the call. Two way voice path is established between PSTN 1 and PSTN 2 PSTN 2 hangs up	PSTN2 should receive Private/Anonymous as CLID	Passed	

g711-016	Call Forwarding Busy Activate	PBX User dials *90 Enter the CFB Destination TN PSTN calls PBX User with CFB	PBX User 1 Dials *90 Announcement is heard PBX User enter PBX User 2 TN Announcement is heard PBX Receives a Bye Busy PBX User 1 PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 does not ring PBX User 2 gets ringing PBX user 2 receives Caller ID (PSTN Originator Caller) PBX User answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed	
g711-017	Call Forwarding Busy Interrogate	PBX User with CFB dials *67* Announcement received	PBX User 1 Dials *67* Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	
g711-018	Call Forwarding Busy Deactivate	PBX User with CFB dials *91 PSTN Calls PBX User	PBX User 1 Dials *91 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	

g711-019	Call Forwarding Busy to Voicemail Activate	PBX User Dials *40 PSTN Dials PBX User with CFB Verify Call goes to Voicemail	PBX User 1 Dials *40 Announcement is received When announcement completes PBX User receives a Bye Busy PBX User 1 PSTN User Calls PBX User 1 Call should go directly to voicemail Announcement is Heard Leave voicemail After leaving voicemail PSTN should receive a Bye PBX User 1 should receive and MWI	Passed	
g711-020	Call Forwarding Busy to Voicemail Deactivate	PBX User with CFB dial #40 PSTN dials PBX User verify Phone rings	PBX User 1 Dials #40 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	

g711-021	Call Forwarding No Answer Activate	PBX User dials *92 Enter the CFNA Destination TN PSTN calls PBX User with CFNA	PBX User 1 Dials *92 Announcement is heard PBX User enters PBX User 2 TN Announcement is heard PBX Receives a Bye PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX User 1 receives Caller ID After timer is RNA is received PBX User 2 gets ringing PBX user 2 receives Caller ID PBX User 2 answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed	

g711-022	Call Forwarding No Answer-RNA Timer	PBX User dials *610 Enter 1 # PSTN calls PBX User with CFNA Verify Call is forwarded	PBX User 1 Dials *610 Announcement is Heard PBX User enter 1 for amount of Rings After announcement completes PBX User 1 receives a Bye PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX User 1 receives Caller ID After timer is RNA is received PBX User 2 gets ringing PBX user 2 receives Caller ID PBX User 2 answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed	Minimum is 0 or 2 rings which can be entered
g711-023	Call Forwarding No Answer Interrogate	PBX User with CFNA dials *61* Announcement received	PBX User 1 Dials *61* Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	
g711-024	Call Forwarding No Answer Deactivate	PBX User with CFNA dials *93 PSTN Calls PBX User	PBX User 1 Dials *93 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	

g711-025	Call Forwarding No Answer to Voicemail Activate	PBX User Dials *41 PSTN Dials PBX User with CFNA Verify Call goes to Voicemail	PBX User 1 Dials *41 Announcement is received When announcement completes PBX User receives a Bye Busy PBX User 1 PSTN User Calls PBX User 1 Call should go to voicemail after RNA timer is reached Announcement is Heard Leave voicemail After leaving voicemail PSTN should receive a Bye PBX User 1 should receive and MWI	Passed	
g711-026	Call Forwarding No Answer to Voicemail Deactivate	PBX User with CFNA dial #41 PSTN dials PBX User verify Phone rings	PBX User 1 Dials #41 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	

g711-027	Call Forwarding Not Reachable Activate	PBX User dials *94 Enter the CFNR Destination TN Unregister Pilot TNs PSTN calls PBX User with CFNR Verify Call is forwarded Register Pilot TNs	PBX User 1 Dials *94 Announcement is heard PBX User enter PBX User 2 TN Announcement is heard PBX Receives a Bye Unplug SBC Lan Cable PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 does not ring PSTN User 2 gets ringing PSTN user 2 receives Caller ID (PSTN Originator Caller) PSTN User answers call 2 way Audio PSTN User 1 releases call PSTN User 2 receives a Bye	Passed	
g711-028	Call Forwarding Not Reachable Interrogate	PBX User with CFNR dials *63* Announcement received	PBX User 1 Dials *63* Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	
g711-029	Call Forwarding Not Reachable Deactivate	PBX User with CFNR dials *95 PSTN Calls PBX User	PBX User 1 Dials *95 Announcement is Heard After announcement completes PBX User 1 receives a Bye	Passed	

g711-030	Call Forwarding Selective Activate	Log into Portal and set up Call forward selective User with a PSTN Number PBX User with CFS enters #76 PSTN User calls PBX User with CFS Call should be call forwarded	Log into Portal and set up Call forward selective User with a PSTN Number PBX User with CFS dials #76 Announcement received PBX User receives a Bye From a Selected PSTN Dial PBX User 1 PBX User should not Ring Call should be call forwarded to the CFS Destination PSTN receives Ringback Destination receives Ringing Destination receives Caller ID (Originator PSTN) Destination answers call 2 way Audio PSTN ends the call Destination receives a Bye	Passed	
g711-031	Call Forwarding Selective Deactivate	PBX User with CFS enters #77 PSTN User calls PBX User Call should not be forwarded	PBX User 1 Dials #77 Announcement is heard PBX Receives a Bye PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed	

g711-032	Call Return by PBX User	PBX User dials *69	PSTN 1 Calls PBX User 1 PSTN 1 receives ringback PBX User 1 receives ringing PBX User 1 receives caller ID PBX User 1 answers call 2 way Audio PSTN 1 ends the call PBX User 1 receives a Bye PBX User 1 Dials *69 PBX User receives Ringback PSTN 1 receives Ringing PSTN receives Caller ID PSTN answers 2 way Audio PSTN releases call PBX User 1 receives a Bye	Passed	
g711-033	Consultative Transfer with SIP REFER	PBX User Calls PSTN PBX User transfers PSTN to PSTN2 PBX User has Audio with PSTNs PSTN 1 has MOH PBX User Transfers Call PSTN and PSTN2 now have audio		Not Run	CUCM 10.5 does not support outbound SIP Transfer with Refer method
g711-034	Unattended Transfer with SIP REFER	PBX User Calls PSTN PBX User transfers PSTN to PSTN2 During Ringback PBX User transfers PSTN 1 has MOH PSTN2 answers call PSTN and PSTN2 now have audio		Not Run	CUCM 10.5 does not support outbound SIP Transfer with Refer method

g711-035	Consultative Transfer	PBX User Calls PSTN PBX User transfers PSTN to PBX User 2 PBX User 1 has Audio with PBX User 2 PSTN 1 has MOH PBX User Transfers Call PSTN and PBX 2 now have audio	PBX User 1 Calls PSTN PBX User receives Ringback PSTN 1 receives Ringing PSTN 1 receives Caller ID PSTN 1 answers 2 way Audio PBX User transfers call to PBX User 2 PSTN User gets MOH PBX User 1 gets Dial tone PBX User 1 dials PBX User 2 Extension PBX User 1 receives Ringback PBX User 2 receives Ringing PBX User 2 receives Caller ID of PBX User 1 PBX User 2 answers the Call 2 way Audio PBX User 1 transfers the call MOH Ends PSTN 1 and PBX User 2 are now connected 2 Way Audio PSTN 1 Ends the call PBX User 2 receives the Bye	Passed	

g711-036	Unattended Transfer	PBX User Calls PSTN PBX User transfers PSTN to PBX User 2 During Ringback PBX User transfers PSTN 1 has MOH PBX User 2 answers call PSTN and PBX User 2 now have audio	PBX User 1 Calls PSTN PBX User receives Ringback PSTN 1 receives Ringing PSTN 1 receives Caller ID PSTN 1 answers 2 way Audio PBX User transfers call to PBX User 2 PSTN User gets MOH PBX User 1 gets Dial tone PBX User 1 dials PBX User 2 Extension PBX User 1 receives Ringback PBX User 2 receives Ringing PBX User 2 receives Caller ID of PSTN 1 PBX User 1 release call PBX User 2 answers the Call MOH Ends 2 way Audio PSTN 1 release the call PBX User 2 receives the Bye	Passed	

g711-037	Call Waiting Persistent Activate	PBX User dials *43 PSTN Calls PBX User PSTN 2 Calls PBX User Verify Call Waiting Tone	PBX User 1 Dials *43 Announcement is heard PBX Receives a Bye PSTN User 1 Calls PBX User 1 PSTN User 1 receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN User 2 Calls PBX User 1 PSTN User 2 receives ringback PBX User 1 receives caller ID PBX User 1 hear Call Waiting Tone PBX User Places PSTN User 1 on Hold PSTN User 1 hears MOH PBX User 1 answers Call from PSTN 2 2 way Audio Verify PBX User 1 can swap between to callers While on PBX User 1 and PSTN User 1 PSTN 1 releases Call PBX User 1 receives a Bye Call 2 should still be up with PSTN 2 hearing MOH	Passed	

g711-038	Call Waiting Persistent Deactivate	PBX User Dials #43 PSTN Calls PBX User PSTN 2 Calls PBX User Call 2 should go to voicemail	PBX User 1 Dials #43 Announcement is heard PBX User 1 Receives a Bye after Announcement is completed PSTN User Calls PBX User 1 PSTN User 1 Calls PBX User 1 PSTN User 1 receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User 1 answers call 2 way Audio PSTN User 1 releases the call PBX User 1 receives a Bye	Passed	Call Forwarding Busy to Voicemail is activated to send PSTN User 2 to voicemail
g711-039	Customer Originated Trace	PSTN Calls PBX User PBX User Answers the Call PBX User Hangs up call PBX User enters *57 Verify announcement	PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN releases call PBX User 2 receives a Bye PBX User 1 Dial *57 Announcement received Announcement Completes PBX User receives a Bye	Passed	
g711-040	Enhanced Call Logs	Log into portal and verify Call logs	Log into the portal for PBX User 1 On main screen verify calls Logs are displayed Missed Received Placed	Passed	

g711-041	Last Number Redial	PBX User dials *66 The last number dialed should be called	PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN releases call PBX User 1 receives a Bye PBX User 1 Dial *66 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN releases call PBX User 1 receives a Bye	Passed	
g711-042	MOH	Verify MOH for conference, transfer, and hold	PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PBX User 1 Places call on Hold PSTN receives MOH PBX User retrieves call from Hold 2 way Audio PSTN releases call PBX User 2 receives a Bye	Passed	

g711-043	Remote Office - Like CFA	Provision Remote office for a SIP Trunk user on the BroadWorks portal to use PSTN number A. Place a call from a PSTN number B to the SIP Trunk user's DID and verify that it is forwarded to PSTN number A (the destination configured in BroadWorks).	Log into the portal for PBX User 1 Set up remote Office to PSTN Number 1 PSTN User 2 Calls PBX User 1 PSTN 2 receives ringback PSTN User 1 gets ringing with PSTN 2 Caller ID and Diversion header for PBX User1 PSTN User 1 answers call 2 way Audio PSTN 1 releases call	Passed	
g711-044	Remote Office - Quick Call	Provision Remote office for a SIP Trunk user 1 on the BroadWorks portal to use PSTN number A. On the BW portal, Manage Users, select Configure Features of User 1, under Quick Call, add PSTN B number and click on the Call Button. PSTN A should Start Ringing with PBX User 1 Caller ID.	Log into the portal for PBX User 1 Set up remote Office to PSTN Number 1 Initiate a Quick Call to PSTN 2 on the portal PSTN User 1 gets ringing with PBX User 1 Caller ID PSTN user 1 answers the call. Now PSTN2 should start ringing with PBX User1 as Caller ID. PSTN 1 might hear ringback based on how long PSTN 2 rings. PSTN 2 answers the call 2 way Audio PSTN 1 releases call	Passed	

g711-045	Remote Office - Click to Call	Provision Remote office for a SIP Trunk user 1 on the BroadWorks portal to use PSTN number A. On the BW portal, Manage Users, select Configure Features of User 1, under Call Logs, select either incoming/outgoing/missed calls and Click on a Call under Phone Number Click To call column. PSTN A should Start Ringing with PBX User 1 Caller ID.	Log into the portal for PBX User 1 Set up remote Office to PSTN Number 1 Review call logs and identify a call log that needs to be called via Click to Call. Click on the identified call log under Click to Call PSTN User 1 gets ringing with PBX User 1 Caller ID PSTN user 1 answers the call. Now PSTN2 should start ringing with PBX User1 as Caller ID. PSTN 1 might hear ringback based on how long PSTN 2 rings. PSTN 2 answers the call 2 way Audio PSTN 1 releases call	Passed	
g711-046	Selective Call Acceptance	Provision selective call acceptance in the BroadWorks portal. Place a call from an accepted TN to the SIP Trunk User. Verify that the call completes normally. Place a call from a TN that is not on the accept list and verify that the call does not reach the SBC.	Log into the portal for PBX User 1 Set up Selected Call Acceptance to PSTN Number 1 PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN releases Call PBX User 1 receives a Bye	Passed	

g711-047	Selective Call Rejection	Provision selective call rejection in the BroadWorks portal. Place a call from a TN not on the reject list to the SIP Trunk User. Verify that the call completes normally. Place a call from a TN that is on the reject list and verify that the call does not reach the SBC.	Log into the portal for PBX User 1 Set up Selected Call rejection to PSTN Number 1 PSTN Calls PBX User 1 Verify PSTN gets an announcement PSTN receives a Bye	Passed	
g711-048	Sequential Ring	Provision sequential ring in the BroadWorks portal. Place a call to the SIP trunk user. Verify that the numbers in the sequential ring list are dialed in order.	Log into the Portal for PBX User 1 Set up Sequential Ring with PBX User 2 and PBX User 3 PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID After RNA Timer PBX User 1 receives a Cancel PBX User 2 gets ringing PBX user 2 receives Caller ID After RNA Timer PBX User 1 receives a Cancel PBX User 3 gets ringing PBX user 3 receives Caller ID PBX User 3 answers call 2 way Audio PSTN releases Call PBX User 3 receives a Bye	Passed	

g711-049	Simultaneous Ring	Provision Simultaneous ring in the BroadWorks portal. Place a call to the SIP trunk user. Verify that the numbers in the Simultaneous ring list are dialed at once.	Log into the Portal for PBX User 1 Set up Simultaneous Ring with PBX User 2 and PBX User 3 PSTN Calls PBX User 1 PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User 2 gets ringing PBX user 2 receives Caller ID PBX User 3 gets ringing PBX user 3 receives Caller ID PBX User 3 Answers Call PBX User 1 and 2 receive a Cancel 2 way Audio PSTN releases Call PSTN User 3 receives a Bye	Passed	
g711-050	Third Party MWI Control NOTIFY	Provision Third Party MWI in the BroadWorks portal. Provision the CT Voice Mail system to notify BroadWorks of unread messages in the user's voice mail box. Confirm that the NOTIFY is sent to BroadWorks and that the NOTIFY is sent to the PBX.	PSTN User Calls PBX User 1 Call should go to voicemail after RNA timer is reached Announcement is Heard Leave voicemail After leaving voicemail PSTN should receive a Bye PBX User 1 should receive and MWI PBX User 1 dials *86 Log into Mailbox Listen To Voicemail Delete Voicemail Verify MWI is gone PBX User 1 ends the Call	Passed	

g711-051	Voice Mail Consultation	Provision Voice Mail n the BroadWorks or NYMPH portal. Provision the PBX to forward calls to an external voice mail system as the user's call coverage. Confirm the PBX user's capability to retrieve voice mail from the external Voice Mail system.	PSTN User Calls PBX User 1 Call should go to voicemail after RNA timer is reached Announcement is Heard Leave voicemail After leaving voicemail PSTN should receive a Bye PBX User 1 should receive and MWI PBX User 1 dials *86 Log into Mailbox Listen To Voicemail Delete Voicemail Verify MWI is gone PBX User 1 ends the Call	Passed	

g711-052	PBX Initiate Conference	PBX User Calls PSTN PBX User Conferences PBX User 2	PBX User 1 Calls PSTN PBX User receives Ringback PSTN 1 receives Ringing PSTN 1 receives Caller ID PSTN 1 answers 2 way Audio PBX User conferences call to PBX User 2 PSTN User gets MOH PBX User 1 gets Dial tone PBX User 1 dials PBX User 2 Extension PBX User 1 receives Ringback PBX User 2 receives Ringing PBX User 2 receives Caller ID of PBX User 1 PBX User 2 answers the Call 2 way Audio PBX User 1 conferences the call MOH Ends PSTN 1, PBX User 1 and PBX User 2 are now connected 2 Way Audio PBX User 1 Ends the call PBX User 2 and PSTN receives the Bye	Passed	PBX user 2 and PSTN still have an audio on after PBX user 1 ends the call

g711-053	PSTN Initiate Conference	PBX User calls PSTN PSTN conferences PBX User2	PBX User 1 Calls PSTN PBX User receives Ringback PSTN 1 receives Ringing PSTN 1 receives Caller ID PSTN 1 answers 2 way Audio PSTN User 1 conferences call to PBX User 2 PBX User 1 gets MOH PSTN User 1 gets Dial tone PSTN User 1 dials PBX User 2 Extension PSTN User 1 receives Ringback PBX User 2 receives Ringing PBX User 2 receives Caller ID of PSTN User 1 PBX User 2 answers the Call 2 way Audio PSTN User 1 conferences the call MOH Ends PSTN 1, PBX User 1 and PBX User 2 are now connected 2 Way Audio PSTN User 1 Ends the call PBX User 1 and PBX User 2 Still Have Audio PBX User 1 End the Call PBX User 2 receives a Bye	Passed	

g711-054	Huntgroup Seq Ring	PSTN Calls Huntgroup Seq ring Answer call on 2nd Member	Log into Admin Portal Create Huntgroup with 3 members PSTN Calls Huntgroup PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID After RNA Timer is reached PBX User 2 gets ringing PBX user 2 receives Caller ID PBX User 2 Answers the call 2 way Audio PSTN ends the call PBX User 2 receives a Bye		Hunt group is not purchased

g711-055	Huntgroup Seq Ring RNA to Voicemail	PSTN calls Huntgroup Seq ring RNA to Voicemail	Log into Admin Portal Create Huntgroup with 3 members PSTN Calls Huntgroup PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID After RNA Timer is reached PBX User 2 gets ringing PBX user 2 receives Caller ID After RNA Timer is reached PBX User 3 gets ringing PBX user 3 receives Caller ID Call should go to voicemail after RNA timer is reached Announcement is Heard Leave voicemail After leaving voicemail PSTN should receive a Bye PBX User 1 should receive and MWI PBX User 1 dials *86 Enter *# Log into HuntGroup Mailbox Listen To Voicemail Delete Voicemail PBX User 1 ends the Call	Not Run	Hunt group is not activated for test account

g711-056	Huntgroup Sim Ring	PSTN calls Huntgroup Sim ring 3 members Answer Call	Log into Admin Portal Create Huntgroup with 3 members with Sequential Ring PSTN Calls Huntgroup PSTN receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User 2 gets ringing PBX user 2 receives Caller ID PBX user 3 receives Caller ID PBX User 3 Answers the call PBX User 3 Answers the Call 2 way Audio PSTN ends the call PBX User 2 receives a Bye	Not Run	Hunt group is not activated for test account
g711-057	PBX to PBX	PBX User Calls PBX User2 Same Trunk Verify RTP is dropped to SBC	PBX User 1 Calls PBX User 2 PBX User 1 receives ringback PBX User 2 gets ringing PBX user 2 receives Caller ID PBX User 2 answers call 2 way Audio RTP is on SBC/PBX PBX User 1 End the call PBX User 2 receives a Bye	Passed	
g711-058	PSTN to PBX	PSTN to PBX User	PSTN User 1 Calls PBX User 1 PSTN User 1 receives ringback PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User answers call 2 way Audio PSTN User Ends The Call PBX User 1 receives a Bye	Passed	

g711-059	PBX to PSTN	PBX User to PSTN	PBX User 1 Calls PSTN User 1 PBX User 1 receives ringback PSTN User 1 gets ringing PSTN user 1 receives Caller ID PSTN User answers call 2 way Audio PSTN User Ends The Call PBX User 1 receives a Bye	Passed	
g711-060	PBX to PBX Different PBX (diff realm)	PBX User to PBX User Different PBX (diff realm)	PBX User 1 Calls PBX User 2 Diff Realm PBX User 1 receives ringback PBX User 2 gets ringing PBX user 2 receives Caller ID PBX User 2 answers call 2 way Audio RTP PBX User 1 End the call PBX User 2 receives a Bye	Passed	
g711-061	PSTN to PBX -Passthrough	PSTN to PBX User Fax Call	PSTN User 1 Fax Calls PBX User 1 Fax PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User 1 Fax answers call Fax is received PBX User Ends The Call PSTN User 1 receives a Bye	Passed	

g711-062	PBX to PSTN -Passthrough	PBX User to PSTN Fax Call	PBX User 1 Fax Calls PSTN User 1 Fax PSTN User 1 gets ringing PSTN user 1 receives Caller ID PSTN User 1 Fax answers call Fax is received PSTN User Ends The Call PBX User 1 receives a Bye	Passed	
g711-063	PSTN to PBX -T38	PSTN to PBX User Fax Call	PSTN User 1 Fax Calls PBX User 1 Fax PBX User 1 gets ringing PBX user 1 receives Caller ID PBX User 1 Fax answers call Fax is received PBX User Ends The Call PSTN User 1 receives a Bye	Passed	
g711-064	PBX to PSTN -T38	PBX User to PSTN Fax Call	PBX User 1 Fax Calls PSTN User 1 Fax PSTN User 1 gets ringing PSTN user 1 receives Caller ID PSTN User 1 Fax answers call Fax is received PSTN User Ends The Call PBX User 1 receives a Bye	Passed	
g711-065	PBX to PSTN - Packet Marking for SIG packets	PBX to PSTN Call to verify that signaling packets are marked with DSCP = 24 (0x18)	All outgoing SIP Signaling packets are marked with DSCP=24	Passed	Same as g711-059
g711-066	PBX to PSTN - Packet Marking for RTP packets	PBX to PSTN Call to verify that rtp packets are marked with DSCP = 40 (0x28)	All outgoing SIP RTP packets are marked with DSCP=40	Passed	Same as g711-059

g711-067	PBX to PSTN - Directory assistance	PBX User Calls PBX 411 and speaks with directory assistant	PBX User 1 dials 411 Call is delivered to Directory Assistant for enquiry Once the user hears an announcement or speaks with an operator, PBX user hangs up the call	Passed	
g711-068	PBX to PSTN - Toll Free	PBX User Calls 800.366.8201 to test toll free numbers	PBX User 1 dials 800.366.8201 (CTL Support) Call is delivered to CenturyLink Support Once the user hears an announcement or speaks with an operator, PBX user hangs up the call	Passed	
g711-069	PBX to PSTN - 911	PBX User Calls 911 to get emergency support	PBX User 1 dials xxx-xxx-xxxx (CTL Rep) Call is delivered to CenturyLink Rep PBX User makes conferences 911 operator PBX User, CTL rep and 911 operator are conferenced ????	Passed	Same routing as for g711-067
g711-070	PBX to PSTN - International	PBX User Calls international number	International Call is successfully established and torn down.	Passed	

S.No	Title	Description	Test Setup	Result	Comment
g729-001	Configure Dual Trunk on PBX	PBX is configured and connected to 2 PSTN GW/SBCs	The steps will be based on the type of PBX being utilized. Ensure that trunks are configured between PBX and SBC. Verify OPTIONS msgs from either PBX or SBC are being responded correctly by the other entity	Passed	

g729-002	Configure Dual Trunk on ITSP	ITSP is configured and connected to 2 PSTN GW/SBCs	The steps will be based on the type of SBC being utilized. Ensure the TWO SBCs are configured with individual trunks to ITSP	Passed	
g729-003	Regitration of Dual Trunks	Ensure that both trunks to ITSP are registered successfully using the individual trunk registration information	1. Each SBC is configured with a trunk to ITSP and associated authentiation/digest and registration information. 2. Invoke a command on SBC to register the trunk with ITSP. 3. Verify that 200 OK is received from ITSP for both the trunks.	Passed	

g729-004	Inbound PSTN calls pick correct trunk to SBC	Verify that PSTN to PBX inbound calls arrive on both the trunks when multiple calls are made	1. Dial an inbound call to the PBX. 2. Verify ringing is heard by calling and called parties. 3. Verify the trace shows a valid ringing indication message 4. Take called party phone off-hook. 5. Verify that a media path is established in both directions. 6. Hang up calling party 7. Verify the IP/PBX receives a Bye message. 8. Make a note of the Trunk on which the call arrived to the SBC and PBX. 9. Repeat the above steps 3 more times (total 4 calls). 10. Verify that calls to PBX arrive on both the trunks. 11. Document Test Results. 12. Save Trace.	Passed	

g729-005	PBX calls are delivered to PSTN on both the trunks	Calls from PBX to PSTN are delivered to ITSP/PSTN utilizing both the configured trunks	1. Dial an outbound call from the PBX. 2. Verify ringing is heard by calling and called parties. 3. Verify the trace shows a valid ringing indication message 4. Take called party phone off-hook. 5. Verify that a media path established in both directions. 6. Hang up Calling Party 7. Verify the IP/PBX sends a Bye message. 8. Make a note of the Trunk on which the call was sent to ITSP. 9. Repeat the above steps 3 more times (total 4 calls). 10. Verify that calls from PBX are sent out on both the trunks to ITSP. 11. Verify each call has PAI sent per the trunk configuration 12. Document Test Results. 13. Save Trace.	Passed	

g729-006	Alien TN calls on 1st trunk	Verify calls are successful with Alien TNs on 1st trunk	1. After Alien TN is set up on a Trunk1 in CenturyLink Network 2. PBX User Places a Call to PSTN 3. PBX User receives ring back 4. PSTN receives ringing 5. PSTN receives caller id of the Alien TN 6. PSTN answers the call 7. 2 way audio is received 8. PBX Phone releases Calls 9. PSTN receives a Bye	Passed	
g729-007	Alien TN calls on 2nd trunk	Verify calls are successful with Alien TNs on 2nd trunk	1. After Alien TN is set up on a Trunk2 in CenturyLink Network 2. PBX User Places a Call to PSTN 3. PBX User receives ring back 4. PSTN receives ringing 5. PSTN receives caller id of the Alien TN 6. PSTN answers the call 7. 2 way audio is received 8. PBX Phone releases Calls 9. PSTN receives a Bye	Passed	

g729-008	Failover of 1st trunk WAN - PSTN-PBX	Ensure that calls are delivered from PSTN to PBX when the first trunk has failed on the WAN side	1. Down the WAN interface associated with Trunk 1. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that all 3 calls are delivered to PBX utilizing Trunk 2 4. PBX user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	
g729-009	Failover of 1st trunk WAN - PBX-PSTN	Ensure that calls are delivered from PBX to PSTN when the first trunk has failed on the WAN side	1. Down the WAN interface associated with Trunk 1. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that all 3 calls are delivered to PSTN utilizing Trunk 2 4. PSTN user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	

g729-010	Restore 1st trunk WAN: PSTN-PBX	Ensure that calls are delivered from PSTN to PBX when the first trunk has has been restored	1. WAN interface associated with Trunk 1 is brought back into service. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that at least one call is delivered to the PBX via Trunk 1 4. PBX user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	
g729-011	Restore 1st trunk WAN: PBX-PSTN	Ensure that calls are delivered from PBX to PSTN when the first trunk has has been restored	1. WAN interface associated with Trunk 1 is brought back into service. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that at least one call is delivered to the PSTN via Trunk 1 4. PSTN user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	

g729-012	Failover of 2nd trunk WAN: PSTN-PBX	Ensure that calls are delivered from PSTN to PBX when the second trunk has failed on the WAN side	1. Down the WAN interface associated with Trunk 2. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that all 3 calls are delivered to PBX utilizing Trunk 1 4. PBX user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	
g729-013	Failover of 2nd trunk WAN: PBX-PSTN	Ensure that calls are delivered from PBX-PSTN when the second trunk has failed on the WAN side	1. Down the WAN interface associated with Trunk 2. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that all 3 calls are delivered to PBX utilizing Trunk 1 4. PSTN user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	

g729-014	Restore 2nd trunk WAN: PSTN-PBX	Ensure that calls are delivered from PSTN to PBX when the second trunk has has been restored	1. WAN interface associated with Trunk 2 is brought back into service. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that at least one call is delivered to the PBX via Trunk 2 4. PSTN user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	
g729-015	Restore 2nd trunk WAN: PBX-PSTN	Ensure that calls are delivered from PBX to PSTN when the second trunk has has been restored	1. WAN interface associated with Trunk 2 is brought back into service. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that at least one call is delivered to the PBX via Trunk 2 4. PBX user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	

g729-016	Failover of 1st trunk LAN - PBX to PSTN	Ensure that calls are delivered from PBX to PSTN when the first trunk has failed on the LAN side	1. Down the LAN interface associated with Trunk 1. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that all 3 calls are delivered to PSTN utilizing Trunk 2 4. PSTN user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	
g729-017	Failover of 1st trunk LAN - PSTN to PBX	Ensure that calls are delivered from PSTN to PBX when the first trunk has failed on the LAN side	1. Down the LAN interface associated with Trunk 1. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that all 3 calls are delivered to PBX utilizing Trunk 2 4. PBX user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	

g729-018	Restore 1st trunk LAN - PBX to PSTN	Ensure that calls are delivered from PBX to PSTN when the first trunk has has been restored	1. LAN interface associated with Trunk 1 is brought back into service. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that at least one call is delivered to the PSTN via Trunk 2 4. PSTN user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	
g729-019	Restore 1st trunk LAN - PSTN to PBX	Ensure that calls are delivered from PSTN to PBX when the first trunk has has been restored	1. LAN interface associated with Trunk 1 is brought back into service. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that at least one call is delivered to the PBX via Trunk 2 4. PBX user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	

g729-020	Failover of 2nd trunk LAN - PBX to PSTN	Ensure that calls are delivered from PBX to PSTN when the second trunk has failed on the LAN side	1. Down the LAN interface associated with Trunk 2. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that all 3 calls are delivered to PSTN utilizing Trunk 1 4. PSTN user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	
g729-021	Failover of 2nd trunk LAN - PSTN to PBX	Ensure that calls are delivered from PSTN to PBX when the second trunk has failed on the LAN side	1. Down the LAN interface associated with Trunk 2. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that all 3 calls are delivered to PBX utilizing Trunk 1 4. PBX user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	

g729-022	Restore 2nd trunk LAN - PBX to PSTN	Ensure that calls are delivered from PBX to PSTN when the second trunk has has been restored	1. LAN interface associated with Trunk 2 is brought back into service. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that at least one call is delivered to the PSTN via Trunk 2 4. PSTN user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	
g729-023	Restore 2nd trunk LAN - PSTN to PBX	Ensure that calls are delivered from PSTN to PBX when the second trunk has has been restored	1. LAN interface associated with Trunk 2 is brought back into service. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that at least one call is delivered to the PBX via Trunk 2 4. PBX user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	

g711-001	Configure Dual Trunk on PBX	PBX is configured and connected to 2 PSTN GW/SBCs	The steps will be based on the type of PBX being utilized. Ensure that trunks are configured between PBX and SBC. Verify OPTIONS msgs from either PBX or SBC are being responded correctly by the other entity	Passed	
g711-002	Configure Dual Trunk on ITSP	ITSP is configured and connected to 2 PSTN GW/SBCs	The steps will be based on the type of SBC being utilized. Ensure the TWO SBCs are configured with individual trunks to ITSP	Passed	
g711-003	Regitration of Dual Trunks	Ensure that both trunks to ITSP are registered successfully using the individual trunk registration information	1. Each SBC is configured with a trunk to ITSP and associated authentiation/digest and registration information. 2. Invoke a command on SBC to register the trunk with ITSP. 3. Verify that 200 OK is received from ITSP for both the trunks.	Passed	

g711-004	Inbound PSTN calls pick correct trunk to SBC	Verify that PSTN to PBX inbound calls arrive on both the trunks when multiple calls are made	1. Dial an inbound call to the PBX. 2. Verify ringing is heard by calling and called parties. 3. Verify the trace shows a valid ringing indication message 4. Take called party phone off-hook. 5. Verify that a media path is established in both directions. 6. Hang up calling party 7. Verify the IP/PBX receives a Bye message. 8. Make a note of the Trunk on which the call arrived to the SBC and PBX. 9. Repeat the above steps 3 more times (total 4 calls). 10. Verify that calls to PBX arrive on both the trunks. 11. Document Test Results. 12. Save Trace.	Passed	

g711-005	PBX calls are delivered to PSTN on both the trunks	Calls from PBX to PSTN are delivered to ITSP/PSTN utilizing both the configured trunks	1. Dial an outbound call from the PBX. 2. Verify ringing is heard by calling and called parties. 3. Verify the trace shows a valid ringing indication message 4. Take called party phone off-hook. 5. Verify that a media path established in both directions. 6. Hang up Calling Party 7. Verify the IP/PBX sends a Bye message. 8. Make a note of the Trunk on which the call was sent to ITSP. 9. Repeat the above steps 3 more times (total 4 calls). 10. Verify that calls from PBX are sent out on both the trunks to ITSP. 11. Verify each call has PAI sent per the trunk configuration 12. Document Test Results. 13. Save Trace.	Passed	

g711-006	Alien TN calls on 1st trunk	Verify calls are successful with Alien TNs on 1st trunk	1. After Alien TN is set up on a Trunk1 in CenturyLink Network 2. PBX User Places a Call to PSTN 3. PBX User receives ring back 4. PSTN receives ringing 5. PSTN receives caller id of the Alien TN 6. PSTN answers the call 7. 2 way audio is received 8. PBX Phone releases Calls 9. PSTN receives a Bye	Passed	
g711-007	Alien TN calls on 2nd trunk	Verify calls are successful with Alien TNs on 2nd trunk	1. After Alien TN is set up on a Trunk2 in CenturyLink Network 2. PBX User Places a Call to PSTN 3. PBX User receives ring back 4. PSTN receives ringing 5. PSTN receives caller id of the Alien TN 6. PSTN answers the call 7. 2 way audio is received 8. PBX Phone releases Calls 9. PSTN receives a Bye	Passed	

g711-008	Failover of 1st trunk WAN - PSTN-PBX	Ensure that calls are delivered from PSTN to PBX when the first trunk has failed on the WAN side	1. Down the WAN interface associated with Trunk 1. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that all 3 calls are delivered to PBX utilizing Trunk 2 4. PBX user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	
g711-009	Failover of 1st trunk WAN - PBX-PSTN	Ensure that calls are delivered from PBX to PSTN when the first trunk has failed on the WAN side	1. Down the WAN interface associated with Trunk 1. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that all 3 calls are delivered to PSTN utilizing Trunk 2 4. PSTN user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	

g711-010	Restore 1st trunk WAN: PSTN-PBX	Ensure that calls are delivered from PSTN to PBX when the first trunk has has been restored	1. WAN interface associated with Trunk 1 is brought back into service. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that at least one call is delivered to the PBX via Trunk 1 4. PBX user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	
g711-011	Restore 1st trunk WAN: PBX-PSTN	Ensure that calls are delivered from PBX to PSTN when the first trunk has has been restored	1. WAN interface associated with Trunk 1 is brought back into service. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that at least one call is delivered to the PSTN via Trunk 1 4. PSTN user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	

g711-012	Failover of 2nd trunk WAN: PSTN-PBX	Ensure that calls are delivered from PSTN to PBX when the second trunk has failed on the WAN side	1. Down the WAN interface associated with Trunk 2. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that all 3 calls are delivered to PBX utilizing Trunk 1 4. PBX user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	
g711-013	Failover of 2nd trunk WAN: PBX-PSTN	Ensure that calls are delivered from PBX-PSTN when the second trunk has failed on the WAN side	1. Down the WAN interface associated with Trunk 2. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that all 3 calls are delivered to PBX utilizing Trunk 1 4. PSTN user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	

g711-014	Restore 2nd trunk WAN: PSTN-PBX	Ensure that calls are delivered from PSTN to PBX when the second trunk has has been restored	1. WAN interface associated with Trunk 2 is brought back into service. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that at least one call is delivered to the PBX via Trunk 2 4. PSTN user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	
g711-015	Restore 2nd trunk WAN: PBX-PSTN	Ensure that calls are delivered from PBX to PSTN when the second trunk has has been restored	1. WAN interface associated with Trunk 2 is brought back into service. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that at least one call is delivered to the PBX via Trunk 2 4. PBX user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	

g711-016	Failover of 1st trunk LAN - PBX to PSTN	Ensure that calls are delivered from PBX to PSTN when the first trunk has failed on the LAN side	1. Down the LAN interface associated with Trunk 1. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that all 3 calls are delivered to PSTN utilizing Trunk 2 4. PSTN user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	
g711-017	Failover of 1st trunk LAN - PSTN to PBX	Ensure that calls are delivered from PSTN to PBX when the first trunk has failed on the LAN side	1. Down the LAN interface associated with Trunk 1. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that all 3 calls are delivered to PBX utilizing Trunk 2 4. PBX user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	

g711-018	Restore 1st trunk LAN - PBX to PSTN	Ensure that calls are delivered from PBX to PSTN when the first trunk has has been restored	1. LAN interface associated with Trunk 1 is brought back into service. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that at least one call is delivered to the PSTN via Trunk 2 4. PSTN user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	
g711-019	Restore 1st trunk LAN - PSTN to PBX	Ensure that calls are delivered from PSTN to PBX when the first trunk has has been restored	1. LAN interface associated with Trunk 1 is brought back into service. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that at least one call is delivered to the PBX via Trunk 2 4. PBX user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	

g711-020	Failover of 2nd trunk LAN - PBX to PSTN	Ensure that calls are delivered from PBX to PSTN when the second trunk has failed on the LAN side	1. Down the LAN interface associated with Trunk 2. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that all 3 calls are delivered to PSTN utilizing Trunk 1 4. PSTN user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	
g711-021	Failover of 2nd trunk LAN - PSTN to PBX	Ensure that calls are delivered from PSTN to PBX when the second trunk has failed on the LAN side	1. Down the LAN interface associated with Trunk 2. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that all 3 calls are delivered to PBX utilizing Trunk 1 4. PBX user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	

g711-022	Restore 2nd trunk LAN - PBX to PSTN	Ensure that calls are delivered from PBX to PSTN when the second trunk has has been restored	1. LAN interface associated with Trunk 2 is brought back into service. 2. Make 3 calls from PBX to PSTN (one after the other) 3. Ensure that at least one call is delivered to the PSTN via Trunk 2 4. PSTN user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	
g711-023	Restore 2nd trunk LAN - PSTN to PBX	Ensure that calls are delivered from PSTN to PBX when the second trunk has has been restored	1. LAN interface associated with Trunk 2 is brought back into service. 2. Make 3 calls from PSTN to PBX (one after the other) 3. Ensure that at least one call is delivered to the PBX via Trunk 2 4. PBX user answers the call 5. Verify two way voice path is established 6. Called Party hangs up 7. Both Calling and Called parties are disconnected 8. Document results 9. Save traces	Passed	

Conclusion

These Application Notes describe the configuration steps required for Sonus SBC 5XX0 Series to successfully interoperate with CenturyLink SIP Trunk. All feature and serviceability test cases were completed and passed with the exceptions/observations noted in Test Results.